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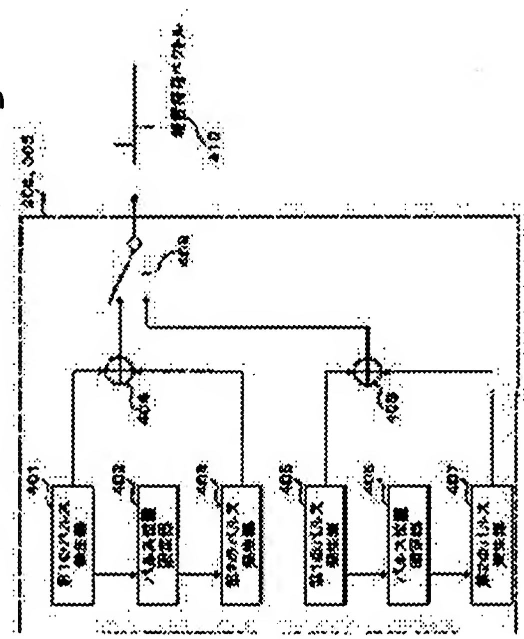
(54) SOUND SOURCE VECTOR GENERATING DEVICE AND VOICE CODING/ DECODING DEVICE

(57)Abstract:

PROBLEM TO BE SOLVED: To improve quality for a silent section and a steady noise section by generating noise code vectors based on first and second pulse positions with two vector generating means, and obtaining a first noise code vector.

SOLUTION: An adder 404 receives a first pulse from a first pulse generator 401 and a second pulse from a second pulse generator 403 and outputs a first noise code vector constituted of two pulses to a selector switch 409. An adder 408 receives a first pulse from a first pulse generator 405 and a second pulse from a second pulse generator 407 and outputs a second noise code vector constituted of two pulses to the selector switch 409. The selector switch 409 selects one of the first noise code vector from the adder 404 and the second noise code vector from the adder 408 and outputs it as a final noise code vector 410.



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DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Field of the Invention] This invention relates to the low bit rate voice to digital converter in the migration communication system which encodes and transmits a sound signal, especially a CELP (Code Excited Linear Prediction) mold voice to digital converter which separates into vocal tract information and excitation information, and expresses a sound signal.
[0002]
[Description of the Prior Art] In the field of digital mobile communication or voice storage, speech information is compressed for a deployment of an electric wave or a storage, and the voice to digital converter for encoding in high efficiency is used. The method which used the CELP (Code Excited Linear Prediction: sign excitation linear predictive coding) method as the base is widely put in practical use in inside and a low bit rate especially. the technique of CELP -- M.R.Schroeder and B.S.Atal: -- "Code-Excited Linear Prediction:(CELP) High-quality Speech at Very Low Bit Rates", Proc.ICASSP-85, 25.1.1, pp.937-940, and 1985" It is shown.
[0003] Voice is performed to a certain fixed frame length (5ms - about 50ms), it performs audio linear prediction for every break and frame, and a CELP mold voice coding method encodes the prediction remainder (excitation signal) by the linear prediction for every frame using the adaptation sign vector and noise sign vector which consist of a known wave. An adaptation sign vector is used choosing it from the adaptation sign book which stores the drive sound-source vector generated in the past, and a noise sign vector is used, choosing it from the noise sign book which stores the vector which has the configuration as which the defined number which was prepared beforehand was determined. The vector generated by arranging how many of that pulse in the vector of a random noise sequence or a different location is used for the noise sign vector stored in a noise sign book.
[0004] An algebraic-sign book is in one of the typical things of the noise sign book of the type which arranges several pulses in a different location. About the algebraic-sign book, concrete contents are shown in "ITU-T recommendation G.729" etc.
[0005] The conventional example of the noise sign vector generation machine using an algebraic-sign book is concretely explained below with reference to drawing 40 .
[0006] Drawing 40 is the fundamental block diagram of the noise sign vector generation machine which used the algebraic-sign book. In drawing, the pulse generated from the 1st pulse generator 1 and the 2nd pulse generator 2 is added with an adder 3, and the noise sign vector is generated by standing two pulses to a different location. The example of an algebraic-sign book is shown in drawing 41 and drawing 42 . An example to which drawing 41 stands two pulses into 80 samples, and drawing 42 show an example which stands three pulses into 80 samples, respectively. In addition, in drawing 41 and drawing 42 , the number indicated by the lower part of a table is the number of the combination of the pulse position.
[0007]
[Problem(s) to be Solved by the Invention] However, in the noise sign vector generation machine using the above-mentioned conventional algebraic-sign book, the retrieval location of each sound-source pulse has been independent, and relative location of a certain sound-source pulse and another sound-source pulse is not used. For this reason, when many numbers of bits are needed and a bias is looked at by the configuration of the noise sign vector which should be generated in order to express sufficient pulse position while it is possible to generate the noise sign vector of various configurations, there is a problem that it is not necessarily an efficient sign book. Moreover, in order to reduce the number of bits required for an algebraic-sign book, the technique of reducing a sound-source pulse number can be considered, but since there are few sound-source pulses in this case, there is a problem that the subjective quality in the

silent section or the stationary noise section deteriorates greatly. Moreover, although there is technique of performing the mode change of a sound source in order to improve the subjective quality of the silent section or the stationary noise section, there is a problem when a mode judging error arises.
[0008] It aims at offering the sound-source vector generation equipment and voice coding / decryption equipment which can improve the coding engine performance to non-vocal sound voice or a background noise, this invention being make in view of this point, being able to reduce the size of a noise sign book, being able to improve the quality over the silent section or the stationary noise section, and suppressing quality degradation at the time of a mode judging error moreover.
[0009]
[Means for Solving the Problem] A pulse-position selection means by which the sound-source vector generation equipment of this invention chooses the 1st pulse position from a predetermined pulse-position candidate, A pulse-position decision means to determine the 2nd pulse position which approaches said 1st pulse position on the basis of said 1st pulse position, The configuration possessing at least two vector generation means to generate a noise sign vector based on said 1st and 2nd pulse positions, and a noise sign vector generation means to acquire the 1st noise sign vector generated by said at least two vector generation means is taken.
[0010] According to this configuration, algebraic-sign book size is efficiently reducible by generating the noise sign vector which has at least two pulses which approach mutually.
[0011] The sound-source vector generation equipment of this invention is set in the above-mentioned configuration, and takes the configuration possessing the control means which controls said pulse-position selection means for the pulse position chosen and determined by said pulse-position selection means and said pulse-position decision means not to come out of a transmission frame.
[0012] According to this configuration, the pulse position chosen and determined by the pulse-position selection means and the pulse-position decision means can search in the pulse-position range which does not come out of a transmission frame, and can generate a noise sign vector.
[0013] The sound-source vector generation equipment of this invention possesses the random sign book which stores the 2nd noise sign vector which includes two or more pulses which do not approach mutually in the above-mentioned configuration, and said noise sign vector generation means takes the configuration which generates a noise sign vector from said 1st and 2nd noise sign vectors.
[0014] According to this configuration, the subjective quality over the silent section or the stationary noise section is improvable by using together the random sign book corresponding to non-vocal sound voice or a stationary noise signal with a partial algebraic-sign book.
[0015] The sound-source vector generation equipment of this invention takes the configuration possessing a mode judging means to judge voice mode, and the number control means of pulse-position candidates which makes the number of said predetermined pulse-position candidates fluctuate according to the judged voice mode in the above-mentioned configuration.
[0016] The coding engine performance to non-vocal sound voice or a background noise is improvable, suppressing quality degradation at the time of a mode judging error by changing the use rate of an algebraic-sign book and a random sign book by mode judging according to this configuration.
[0017] The sound-source vector generation equipment of this invention possesses a PAWA calculation means to compute PAWA of an excitation signal, and an average power calculation means to compute average power when the judged voice mode is noise mode, in the above-mentioned configuration, and said number control means of pulse-position candidates makes the number of said predetermined pulse-position candidates fluctuate based on said average power, and takes a configuration.
[0018] According to this configuration, the coding engine performance to non-vocal sound voice or a background noise is improvable, suppressing more efficiently quality degradation at the time of a mode judging error.
[0019] The voice to digital converter of this invention is characterized by having sound-source vector generation equipment of the above-mentioned configuration.
[0020] The voice to digital converter of this invention An excitation vector generation means to generate a new excitation vector from the adaptation sign vector outputted from the adaptation sign book which stored the excitation vector, and the noise sign vector outputted from the partial algebraic-sign book which stored the noise sign vector acquired by sound-source vector generation equipment according to claim 1, The configuration possessing a renewal means of an excitation vector to update the excitation vector stored in the adaptation sign book to said new excitation vector, and a speech synthesis signal generation means to generate a speech synthesis signal using said new excitation vector and the quantized linear-predictive-coding result is taken.

[0021] According to this configuration, by generating the noise sign vector which has at least two pulses which approach mutually, algebraic-sign book size can be reduced efficiently and a voice to digital converter with small bit rate and amount of operations can be realized.
[0022] A sound-source parameter decode means to decode the sound-source parameter with which the voice decryption equipment of this invention includes the index information which specifies the positional information of an adaptation sign vector, and a noise sign vector, An excitation vector generation means to generate an excitation vector using the noise sign vector which has at least two pulses which are acquired from the adaptation sign vector acquired from the positional information of said adaptation sign vector, and said index information, and which approach mutually, The configuration possessing a renewal means of an excitation vector to update the excitation vector stored in the adaptation sign book to said excitation vector, and a speech synthesis signal generation means to generate a speech synthesis signal using said excitation vector and the decrypted quantization linear-predictive-coding result is taken.
[0023] Since the noise sign vector which has at least two pulses which approach mutually is used according to this configuration, algebraic-sign book size can be reduced efficiently and voice decryption equipment with a small bit rate can be realized.
[0024] The voice coding decryption equipment of this invention generates the sound-source vector which consisted of three sound-source pulses, and takes the configuration possessing the partial algebraic-sign book which stores this sound-source vector, a limit means to restrict so that a sound-source vector with at least 1 set of comparatively narrow sound-source pulse separations may be generated among said sound-source vectors, and the random sign book used accommodative according to the size of said partial algebraic-sign book.
[0025] Since according to this configuration a sound-source pulse is set as three pulses and a partial algebraic-sign book is constituted, the high voice coding decryption equipment of fundamentality ability is realizable.
[0026] The voice coding decryption equipment of this invention takes the configuration with which a limit means classifies according to the location (index) of said sound-source pulse in the above-mentioned configuration.
[0027] According to this configuration, since regular sound-source pulse-position retrieval can be performed, the amount of operations which retrieval takes can be held down to necessary minimum.
[0028] In the above-mentioned configuration, the voice coding decryption equipment of this invention divides a partial algebraic-sign book, and takes the configuration only whose part which reduced the size of said partial algebraic-sign book enlarges the rate of a random sign book.
[0029] According to this configuration, even if it changes random sign book size using mode information etc., it is possible to share the index of an intersection, and the effect of errors, such as mode information, can be suppressed.
[0030] In the above-mentioned configuration, the random sign book consists of two or more channels, and the voice coding decryption equipment of this invention takes the configuration which restricts the location of said sound-source pulse as prevents that a sound-source pulse laps between channels.
[0031] Since the orthogonality between the vectors generated from each channel in a sound-source field can be guaranteed according to this configuration, an efficient random sign book can be constituted.
[0032] The voice coding decryption equipment of this invention takes the configuration possessing the algebraic-sign book which stores a sound-source vector, a diffusion pattern generation means to generate a diffusion pattern according to PAWA of the noise section in voice data, and a pattern diffusion means to diffuse the pattern of the sound-source vector outputted from said algebraic-sign book according to said diffusion pattern.
[0033] According to this configuration, since the noise nature of a diffusion pattern is controllable according to noy ZUPAWA, robust voice coding decryption equipment is realizable to noise level.
[0034] In the above-mentioned configuration, the voice coding decryption equipment of this invention generates a diffusion pattern with high noise nature, when average noy ZUPAWA has a large diffusion pattern generation means, and when PAWA is small, it takes the configuration in which a diffusion pattern with low noise nature is generated.
[0035] According to this configuration, when noise level is high, when noise level is low, a cleaner signal can be expressed for a noise-signal.
[0036] The voice coding decryption equipment of this invention takes the configuration in which a diffusion pattern generation means generates a diffusion pattern according to the mode of voice data in the above-mentioned configuration.
[0037] According to this configuration, according to the mode, it also becomes possible to make noise nature of a diffusion pattern into below whenever [middle], and the voice quality in a noise can be improved in the voice section (voiced section).
[0038] The base station equipment of this invention is characterized by having the voice to digital converter of the above-mentioned configuration. Moreover, the communication terminal of this invention is characterized by having the

voice to digital converter of the above-mentioned configuration.
[0039] The pulse-position selection process that the sound-source vector generation method of this invention chooses the 1st pulse position from a predetermined pulse-position candidate, The pulse-position decision process of determining the 2nd pulse position which approaches said 1st pulse position on the basis of said 1st pulse position, At least two vector generation processes which generate a noise sign vector based on said 1st and 2nd pulse positions, and the noise sign vector generation process of acquiring the 1st noise sign vector generated by said at least two vector generation means are provided.
[0040] According to this approach, algebraic-sign book size is efficiently reducible by generating the noise sign vector which has at least two pulses which approach mutually.
[0041]
[Embodiment of the Invention] Using a partial algebraic-sign book generate [the main point of this invention / a noise sign vector], i.e., when at least two among two or more sound-source pulses generated from an algebraic-sign book use the noise sign vector which generates only combination which approaches, algebraic-sign book size is reduced efficiently. Moreover, the subjective quality over the silent section or the stationary noise section is improved using together the random sign book corresponding to non-vocal sound voice or a stationary noise signal with a partial algebraic-sign book, i.e., by storing a sound-source vector effective in the silent section or the stationary noise section. Furthermore, by the mode judging result, suppressing quality degradation at the time of a mode judging error by switching the ratio of partial algebraic-sign book size and the size of the random sign book used together, the coding engine performance to non-vocal sound voice or a background noise is improved, and subjective quality is improved.
[0042] Here, as for the approaching pulse, the distance from a certain pulse says the thing of the pulse which is below 10 sample extent in the digital signal for 1.25 or less ms, i.e., a 8kHz sampling.
[0043] Hereafter, the gestalt of operation of this invention is explained to a detail with reference to an accompanying drawing.
[0044] (Gestalt 1 of operation) Drawing 1 is the block diagram showing a sound signal transmitter and/or a receiver equipped with voice coding and/or decryption equipment concerning this invention.
[0045] In the sound signal transmitter shown in drawing 1 , a sound signal 101 is changed into an electric analog signal by the audio input unit 102, and is outputted to A/D converter 103 by it. An analog sound signal is changed into a digital sound signal by A/D converter 103, and is outputted to a voice to digital converter 104 by it. A voice to digital converter 104 outputs the information which performed voice coding processing and was encoded to the RF modulator 105. In the RF modulator 105, to the encoded sound signal, processing for sending out as electric waves, such as a modulation, magnification, and sign diffusion, is performed, and the encoded sound signal is outputted to the transmitting antenna 1106. Finally an electric wave (RF signal) is sent out from the transmitting antenna 106.
[0046] On the other hand, in a receiver, a receiving antenna 107 receives an electric wave (RF signal). An input signal is sent to the RF demodulator 108. The RF demodulator 108 performs processing for changing electric-wave signals, such as the sign back diffusion of electrons and a recovery, into encoded information, and outputs encoded information to voice decryption equipment 109. Voice decryption equipment 109 performs decode processing of encoded information, and outputs a digital decode sound signal to D/A converter 110. D/A converter 110 changes into an analog decode sound signal the digital decode sound signal outputted from voice decryption equipment 109, and outputs it to an audio output device 111. Finally an audio output device 111 changes and outputs an electric analog decode sound signal to decode voice.
[0047] Next, the noise sign vector generation machine in the sound signal transmitter and/or receiver which have the above-mentioned configuration is explained. Drawing 2 is the block diagram showing the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 1 of operation. The voice to digital converter shown in this drawing is equipped with the pretreatment machine 201, the LPC analyzer 202, the LPC quantizer 203, the adaptation sign book 204, a multiplier 205, the partial algebraic-sign book 206, a multiplier 207, an adder 208, the LPC composition filter 209, an adder 210, the acoustic-sense weighting machine 211, and the error minimization machine 212.
[0048] In this noise sign vector generation machine, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 201 at every batch time amount (frame). The pretreatment machine 201 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound signal.

[0049] The signal after pretreatment is outputted to the LPC analyzer 202 and an adder 210. The LPC analyzer 202

performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 201, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 203. The LPC quantizer 203 quantizes LPC inputted from the LPC analyzer 202, outputs Quantization LPC to the LPC composition filter 209, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.

[0050] The adaptation sign book 204 is the buffer of the excitation vector (vector outputted from an adder 208) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 212, and outputs it to a multiplier 205. A multiplier 205 multiplies the adaptation sign vector outputted from the adaptation sign book 204 by adaptation sign vector gain, and outputs it to an adder 208. Adaptation sign vector gain is specified with an error minimization vessel. The partial algebraic-sign book 206 is a sign book which has a configuration similar to drawing 4 , drawing 10 , or this which is mentioned later, and outputs the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close] to a multiplier 207.

[0051] A multiplier 207 multiplies the noise sign vector outputted from the partial algebraic-sign book 206 by noise sign vector gain, and outputs it to an adder 208. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 205, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 207, an adder 208 generates an excitation vector and outputs it to the adaptation sign book 204 and the LPC composition filter 209.

[0052] The excitation vector outputted to the adaptation sign book 204 is used when updating the adaptation sign book 204, and the excitation vector outputted to the LPC composition filter 209 is used in order to generate synthesized speech. The LPC composition filter 209 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 203, drives an LPC composition filter using the excitation vector outputted from the adder 208, and outputs a composite signal to an adder 210.

[0053] the difference (error) of the input sound signal after pretreatment to which the adder 210 was outputted from the pretreatment machine 201, and the composite signal outputted from the LPC composition filter 209 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 211. The acoustic-sense weighting machine 211 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 210 as an input, and outputs it to the error minimization machine 212. The error minimization machine 212 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 211 as an input. For example, the location which starts an adaptation sign vector from the adaptation sign book 204 so that the square sum may serve as min, The value of the noise sign vector generated from the partial algebraic-sign book 206, the adaptation sign vector gain by which it multiplies with a multiplier 205, and the noise sign vector gain by which it multiplies with a multiplier 207 is adjusted, each is encoded, and it outputs to a decoder side through a transmission line as sound-source parameter coded data.

[0054] Drawing 3 is the block diagram showing voice decryption equipment equipped with the noise sign vector generation machine concerning the gestalt 1 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 301, the sound-source parameter decoder 302, the adaptation sign book 303, a multiplier 304, the partial algebraic-sign book 305, a multiplier 306, an adder 307, the LPC composition filter 308, and the after-treatment machine 309.

[0055] LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 301 and the sound-source parameter decoder 302 per frame through a transmission line, respectively. The LPC decoder 301 decodes Quantization LPC and outputs it to the LPC composition filter 308. When using Quantization LPC with the after-treatment vessel 309, it is outputted also to the after-treatment machine 309 at coincidence. The sound-source parameter decoder 302 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the adaptation sign book 303, a multiplier 304, the partial algebraic-sign book 305, and a multiplier 306.

[0056] The adaptation sign book 303 is the buffer of the excitation vector (vector outputted from an adder 307) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 302, and outputs it to a multiplier 304. A multiplier 304 multiplies the adaptation sign vector outputted from the adaptation sign book 303 by the adaptation sign vector gain inputted from the sound-source parameter decoder 302, and outputs it to an adder 307.

[0057] The partial algebraic-sign book 305 is the same partial algebraic-sign book as what was shown in 206 of drawing 2 which has a configuration similar to drawing 4 , drawing 10 , or this which is mentioned later, and outputs the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the index inputted from the sound-source parameter decoder 304 is close] to a multiplier 306.

[0058] A multiplier 306 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise

sign vector gain inputted from the sound-source parameter decoder 302, and outputs it to an adder 307. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 306, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 306, an adder 307 generates an excitation vector and outputs it to the adaptation sign book 303 and the LPC composition filter 308.

[0059] The excitation vector outputted to the adaptation sign book 303 is used when updating the adaptation sign book 303, and the excitation vector outputted to the LPC composition filter 308 is used in order to generate synthesized speech. The LPC composition filter 308 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 301, drives an LPC composition filter using the excitation vector outputted from the adder 307, and outputs a composite signal to the after-treatment machine 309.

[0060] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 308 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 309 outputs as decode voice data.

[0061] Next, the noise sign vector generation machine concerning this invention is explained to a detail. Drawing 4 is the block diagram showing the configuration of the noise sign vector generation equipment concerning the gestalt 1 of operation of this invention.

[0062] The 1st pulse generator 401 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 5 (a), and outputs it to an adder 404. Moreover, the 1st pulse generator 401 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 402. The pulse-position limited machine 402 inputs the 1st pulse position from the 1st pulse generator 401, and determines the location candidate of the 2nd pulse on the basis of the location.

[0063] The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 2 of drawing 5 (a). The pulse-position limited machine 402 outputs the location candidate of the 2nd pulse to the 2nd pulse generator 403. The 2nd pulse generator 403 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 402, and outputs it to an adder 404.

[0064] An adder 404 outputs the 1st noise sign vector which inputs the 1st pulse outputted from the 1st pulse generator 401, and the 2nd pulse outputted from the 2nd pulse generator 403, and consists of two pulses to a circuit changing switch 409.

[0065] On the other hand, the 2nd pulse generator 407 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 5 (b), and outputs it to an adder 408. Moreover, the 2nd pulse generator 407 outputs at coincidence the positional information which stood the 2nd pulse to the pulse-position limited machine 406. The pulse-position limited machine 406 inputs the 2nd pulse position from the 2nd pulse generator 407, and determines the location candidate of the 1st pulse on the basis of the location.

[0066] The location candidate of the 1st pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 1 of drawing 5 (b). The pulse-position limited machine 406 outputs the location candidate of the 1st pulse to the 1st pulse generator 405. The 1st pulse generator 405 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 406, and outputs it to an adder 408.

[0067] An adder 408 outputs the 2nd noise sign vector which inputs the 1st pulse outputted from the 1st pulse generator 405, and the 2nd pulse outputted from the 2nd pulse generator 407, and consists of two pulses to a circuit changing switch 409.

[0068] A circuit changing switch 409 chooses one of the 1st noise sign vector outputted from an adder 404, and the 2nd noise sign vector outputted from an adder 408, and outputs it as a final noise sign vector 410. This selection is specified by control from the outside.

[0069] In addition, when the pulse which expresses one side absolutely as mentioned above between two pulses in a location, and is absolutely expressed in a location when another side is expressed as mentioned above in a relative position is near a frame tail, the pulse expressed in a relative position may overflow out of a frame. For this reason, in an actual search algorithm, it is possible to divide into three kinds of retrieval location patterns (a-c), and to search, as only the part which the combination to protrude produces is used as another pattern and shown in drawing 5 . Drawing 5 makes frame length 80 samples (0-79), and an example in the case of standing two pulses into one frame is shown. From the sign book shown in drawing 5 , a part of total entry of a noise sign vector generable from the conventional algebraic-sign book shown in drawing 40 is generable. Suppose that the algebraic-sign book of this invention as shown

in drawing 5 is called a partial algebraic-sign book in this semantics.
[0070] With reference to drawing 6 - drawing 8 , it explains below that processing of the noise sign vector generation method (the coding approach, the noise sign book retrieval approach) in the gestalt of the above-mentioned implementation using the sign book of drawing 5 flows. By drawing 6 , the polarity (+, -) of a pulse shows concretely the case which carries out a thing assumption and encodes only the location of a pulse where it encodes separately.
[0071] First, in step (it abbreviates to ST hereafter) 601, initialization of loop variable i, the error function maximum Max, Index idx, the output index index, the 1st pulse position position1, and the 2nd pulse position position2 is performed.
[0072] Loop variable i is used as a loop variable of the pulse expressed absolutely in a location here, and initial value is 0. The error function maximum Max is initialized by the minimum value (for example, "-10^32") which can be expressed, and it is used in order to maximize the error performance index calculated by the retrieval loop formation. Index idx is an index given to each of the code vector generated by this noise sign vector generation method, initial value is 0, and whenever it changes one location of a pulse, the increment of it is carried out. The location of the 1st pulse where the index of a noise sign vector with which index is finally outputted, and position1 are finally determined, and position2 are the locations of the 2nd pulse finally determined.
[0073] Next, the 1st pulse position (p1) is set to pos1a [j] in ST602. pos1a[ ] is a location (0, 2, .., 72) shown in the column of the pulse number 1 of drawing 5 (a). Here, the 1st pulse is a pulse expressed absolutely in a location.
[0074] Next, initialization of loop variable j is performed in ST PU 603. Loop variable j is the loop variable of the pulse expressed in a relative position, and initial value is 0. Here, the 2nd pulse is expressed in a relative position.
[0075] Next, the 2nd pulse position (p2) is set to p1+pos2a [j] in ST604. p1 is the 1st pulse position already set in ST602, and is pos2a[4] = {1, 3, 5, 7}. The size (the total number of entries of a noise sign vector) of a partial algebraic-sign book can be reduced by reducing the number of elements of pos2a[]. In this case, it is necessary to change the contents of drawing 5 (c) according to the reduced number. Moreover, it is also the same as when increasing.
[0076] Next, in ST605, the error performance index E at the time of standing a pulse to the two set pulse positions is calculated. An error performance index is for evaluating the error of the vector used as a target, and the vector compounded from a noise sign vector, for example, the following formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.
[Equation 1]

$$\frac{(x^{t} H c_i)^2}{c_i^{t} H^{t} H c_i}$$

x: ターゲットベクトル
H: 合成フィルタのインパルス畳み込み行列
c: 雑音符号ベクトル（iはインデックス番号））

[0077] Next, in ST606, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If E value is over Maximum Max, and it progresses to ST607 and has not exceeded, ST607 will be skipped and it will progress to ST608.
[0078] In ST607, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST605, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.
[0079] Next, in ST608, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0080] Next, in ST609, loop variable j checks that it is under total NUM2a of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2a=4. When loop variable j is under NUM2a, in order to repeat the loop formation of j, it returns to ST604. If loop variable j has reached NUM2a, it will end and the loop formation of j will progress to ST610.
[0081] The increment of loop variable i is performed in ST610. By incrementing loop variable i, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0082] Next, in ST611, loop variable i checks that it is under total NUM1a of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1a=37. When loop variable i is under NUM1a, in order

to repeat the loop formation of i, it returns to ST602. If loop variable i has reached NUM1a, it will end and the loop formation of i will progress to ST701 of drawing 7 . When it progresses to ST612, retrieval of drawing 5 (a) is ended and the retrieval loop formation of drawing 5 (b) is started.

[0083] Next, in ST701, loop variable i is cleared and it is set to 0. The 2nd pulse position (p2) is set to pos2b [i] in ST702. pos2b[ ] is a location (1, 3, .., 61) shown in the column of the pulse number 2 of drawing 5 (b). Here, the 2nd pulse is a pulse expressed absolutely in a location.

[0084] Next, initialization of loop variable j is performed in ST703. Loop variable j is the loop variable of the pulse expressed in a relative position, and initial value is 0. Here, the 1st pulse is expressed in a relative position.

[0085] Next, the 1st pulse position (p1) is set to p2+pos1b [j] in ST704. p2 is the 2nd pulse position and pos1b[4] = {1, 3, 5, 7} which have already been set in ST702. The size (the total number of entries of a noise sign vector) of a partial algebraic-sign book can be reduced by reducing the number of elements of pos1b[ ]. In this case, it is necessary to change the contents of drawing 5 (c) according to the reduced number. Moreover, it is also the same as when increasing the number of elements of pos1b[ ].

[0086] Next, in ST705, the error performance index E at the time of standing a pulse to the two set pulse positions is calculated. A formula as an error performance index been for evaluating the error of the vector used as a target and the vector compounded from a noise sign vector, for example, shown in a formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.

[0087] Next, in ST706, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If E value is over Maximum Max, and it progresses to ST707 and has not exceeded, ST707 is skipped and it progresses to ST708.

[0088] In ST707, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST705, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.

[0089] Next, in ST708, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.

[0090] Next, in ST709, loop variable j checks that it is under total NUM1b of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1b=4. When loop variable j is under NUM1b, in order to repeat the loop formation of j, it returns to ST704. If loop variable j has reached NUM1b, it will end and the loop formation of j will progress to ST710.

[0091] The increment of loop variable i is performed in ST701. By incrementing loop variable i, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.

[0092] Next, in ST711, loop variable i checks that it is under total NUM2b of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2b=36. When loop variable i is under NUM2b, in order to repeat the loop formation of i, it returns to ST702. If loop variable i has reached NUM2b, it will end and the loop formation of i will progress to ST801 of drawing 8 . When it progresses to ST801, retrieval of drawing 5 (b) is ended and the retrieval loop formation of drawing 5 (c) is started.

[0093] In ST801, loop variable i is cleared and it is set to 0. Next, the 1st pulse position (p1) is set to pos1c [i] in ST802. pos1c[ ] is a location (74, 76, 78) shown in the column of the pulse number 1 of drawing 5 (c). here -- the -- the pulse of both [2nd] one is expressed absolutely in a location.

[0094] Next, initialization of loop variable j is performed in ST803. Loop variable j is the loop variable of the 2nd pulse, and initial value is 0.

[0095] Next, the 2nd pulse position (p2) is set to pos2c [j] in ST804. pos2c[ ] is a location {73, 75, 77, 79} shown in the column of the pulse number 2 of drawing 5 (c).

[0096] Next, in ST805, the error function E at the time of standing a pulse to the two set pulse positions is calculated. A formula as an error function been for evaluating the error of the vector used as a target and the vector compounded from a noise sign vector, for example, shown in a formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.

[0097] Next, in ST806, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If it has exceeded, and progresses to ST807 and has not exceeded, ST807 is skipped and it progresses to ST808. In ST807, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST805, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.
[0098] Next, in ST808, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0099] Next, in ST809, loop variable j checks that it is under total NUM2c of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2c=4. When loop variable j is under NUM2c, in order to repeat the loop formation of j, it returns to ST804. If loop variable j has reached NUM2c, it will end and the loop formation of j will progress to ST810.
[0100] The increment of loop variable i is performed in ST810. By incrementing loop variable i, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0101] Next, in ST811, loop variable i checks that it is under total NUM1c of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1c=3. When loop variable i is under NUM1c, in order to repeat the loop formation of i, it returns to ST802. If loop variable i has reached NUM1c, it will end and the loop formation of i will progress to ST812. When it progresses to ST812, it ends and all retrieval ends retrieval of drawing 5 (c).
[0102] Finally, in ST812, index which it is as a result of retrieval is outputted. Although it is not necessary to output the two pulse positions position1 and position2 corresponding to index, it can be used for local decode. In addition, since the polarity (are they + or -?) of each pulse can be beforehand determined by doubling with the vector xH in a formula (1) (only the time of the correlation of xH and c in a formula (1) being forward is considered), it is omitted with the gestalt of the above-mentioned implementation.
[0103] With reference to drawing 9 , the flow of processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation using the sign book of drawing 5 is explained below. By drawing 9 , the polarity (+, -) of a pulse shows concretely the case which carries out a thing assumption and decrypts only the location of a pulse where it is decrypted separately.
[0104] First, in ST901, the index index of the noise sign vector received from the encoder confirms whether to be less than one IDX. IDX1 is the sign book size of the part of (a) in the sign book of drawing 5 , and is the value of idx in the time in ST601 of drawing 6 . It is more specifically IDX1=32x4=128. If index is less than one IDX, since the two pulse positions are parts expressed by drawing 5 (a), it progresses to ST602. In order to check further since it becomes drawing 5 (b) or the part of (c) when index(es) are one or more IDX(s), it progresses to ST905.
[0105] It asks for the quotient idx1 which broke index by Num2a in ST902. idx1 serves as an index number of the 1st pulse. In ST902, int() is a function which asks for the integer part in ( ).
[0106] Next, in ST903, idx2 is calculated just because it broke index by Num2a. idx2 serves as an index number of the 2nd pulse.
[0107] Next, in ST904, the location position2 of the 2nd pulse is determined using the sign book of drawing 5 (a) using idx2 asked for the location position1 of the 1st pulse using idx1 calculated by ST902 by ST903, respectively. position1 and position2 which were determined are used by ST914.
[0108] When index(es) are one or more IDX(s) in ST901, it progresses to ST905. In ST905, index confirms whether to be less than two IDX. IDX2 is the sign book size which doubled the part of (a) in the sign book of drawing 5 , and the part of (b), and is the value of idx in the time in ST801 of drawing 6 . It is more specifically IDX2=32x4+31x4=252. If index is less than two IDX, since the two pulse positions are parts expressed by drawing 5 (b), it progresses to ST906. Since it is the part expressed by drawing 5 (c) when index(es) are two or more IDX(s), it progresses to ST910.
[0109] In ST906, IDX1 is subtracted from index and it progresses to ST907. It asks for the quotient idx2 which broke index after IDX1 subtraction by Num1b in ST907. This idx2 serves as an index number of the 2nd pulse. In ST907, int ( ) is a function which asks for the integer part in ( ).
[0110] Next, in ST908, idx1 is calculated just because it broke index after IDX1 subtraction by Num1b. This idx1 serves as an index number of the 1st pulse.
[0111] Next, in ST909, the location position1 of the 1st pulse is determined using the sign book of drawing 5 (b) using idx1 asked for the location position2 of the 2nd pulse using idx2 calculated by ST907 by ST908, respectively. position1 and position2 which were determined are used by ST914.

[0112] When index(es) are two or more IDX(s) in ST905, it progresses to ST910. In ST910, IDX2 is subtracted from index and it progresses to ST911. It asks for the quotient idx1 which broke index after IDX2 subtraction by Num2c in ST911. This idx1 serves as an index number of the 1st pulse. In ST911, int ( ) is a function which asks for the integer part in ( ).

[0113] Next, in ST912, idx2 is calculated just because it broke index after IDX2 subtraction by Num2c. This idx2 serves as an index number of the 2nd pulse.

[0114] Next, in ST913, the location position2 of the 2nd pulse is determined using the sign book of drawing 5 (c) using idx2 asked for the location position1 of the 1st pulse using idx1 calculated by ST911 by ST912, respectively. position1 and position2 which were determined are used by ST914.

[0115] In ST914, noise sign vector code[ ] is generated using the location position1 of the 1st pulse, and the location position2 of the 2nd pulse. That is, the vector which is 0 is generated except code [position1] and code [position2]. code "position1" and code "position2" are set to the polarities [sign / sign and / +1] 1 decoded separately or 1 by two (sign1 and sign2 take the value of +1 or 1). It is the noise sign vector by which code[ ] is decoded.

[0116] Next, the example of a configuration of the partial algebraic-sign book whose pulse number is three is shown in drawing 10 .

[0117] The example of a configuration in drawing 10 takes the configuration which limits a pulse retrieval location so that at least two of three may be arranged in the location which approached. The sign book corresponding to this configuration is shown in drawing 11 .

[0118] Explanation is added to below using drawing 10 . The 1st pulse generator 1001 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (a), and outputs it to an adder 1005. Moreover, the 1st pulse generator 1001 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 1002. The pulse-position limited machine 1002 inputs the positional information of the 1st pulse from the 1st pulse generator 1001, and determines the location candidate of the 2nd pulse on the basis of the location. The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 2 of drawing 11 (a).

[0119] The pulse-position limited machine 1002 outputs the candidate of the 2nd pulse position to the 2nd pulse generator 1003. The 2nd pulse generator 1003 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 1002, and outputs it to an adder 1005. The 3rd pulse generator 1004 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (a), and outputs it to an adder 1005. An adder 1005 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1001, 1003, and 1004, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0120] The 1st pulse generator 1006 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (d), and outputs it to an adder 1010. Moreover, the 1st pulse generator 1006 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 1007. The pulse-position limited machine 1007 inputs the positional information of the 1st pulse from the 1st pulse generator 1006, and determines the location candidate of the 3rd pulse on the basis of the location. The location candidate of the 3rd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 3 of drawing 11 (d).

[0121] The pulse-position limited machine 1007 outputs the candidate of the 3rd pulse position to the 3rd pulse generator 1008. The 3rd pulse generator 1008 stands the 3rd pulse to one of the location candidates of the 3rd pulse inputted from the pulse-position limited machine 1007, and outputs it to an adder 1010. The 2nd pulse generator 1009 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (d), and outputs it to an adder 1010. An adder 1010 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1006, 1008, and 1009, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0122] The 3rd pulse generator 1011 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (b), and outputs it to an adder 1015. The 2nd pulse generator 1012 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (b), and outputs it to an adder 1015. Moreover, the 2nd pulse generator 1012 outputs at coincidence the location which stood the 2nd pulse to the pulse-position limited machine 1013. The pulse-position limited machine 1013 inputs the location of the 2nd pulse from the 2nd pulse generator 1012, and determines the location candidate of the 1st pulse on the basis of the location. The location candidate of the 1st pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 1 of drawing 11 (b).

[0123] The pulse-position limited machine 1013 outputs the location candidate of the 1st pulse to the 1st pulse generator 1014. The 1st pulse generator 1014 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 1013, and outputs it to an adder 1015. An adder 1015 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1011, 1012, and 1014, and outputs the noise sign vector which consists of three pulses to a circuit changing switch 1031.

[0124] The 1st pulse generator 1016 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (g), and outputs it to an adder 1020. The 2nd pulse generator 1017 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (g), and outputs it to an adder 1020. Moreover, the 2nd pulse generator 1017 outputs at coincidence the location which stood the 2nd pulse to the pulse-position limited machine 1018. The pulse-position limited machine 1018 inputs the location of the 2nd pulse from the 2nd pulse generator 1017, and determines the location candidate of the 3rd pulse on the basis of the location. The location candidate of the 3rd pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 3 of drawing 11 (g).

[0125] The pulse-position limited machine 1018 outputs the location candidate of the 3rd pulse to the 3rd pulse generator 1019. The 3rd pulse generator 1019 stands the 3rd pulse to one of the location candidates of the 3rd pulse inputted from the pulse-position limited machine 1018, and outputs it to an adder 1020. An adder 1020 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1016, 1017, and 1019, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0126] The 2nd pulse generator 1021 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (e), and outputs it to an adder 1025. The 3rd pulse generator 1024 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (e), and outputs it to an adder 1025. Moreover, the 3rd pulse generator 1024 outputs at coincidence the location which stood the 3rd pulse to the pulse-position limited machine 1023. The pulse-position limited machine 1023 inputs the location of the 3rd pulse from the 3rd pulse generator 1024, and determines the location candidate of the 1st pulse on the basis of the location. The location candidate of the 1st pulse is expressed with the relative expression from the location (= P3) of the 3rd pulse as shown in the column of the pulse number 1 of drawing 11 (e).

[0127] The pulse-position limited machine 1023 outputs the location candidate of the 1st pulse to the 1st pulse generator 1022. The 1st pulse generator 1022 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 1023, and outputs it to an adder 1025. An adder 1025 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1021, 1022, and 1024, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0128] The 1st pulse generator 1026 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (h), and outputs it to an adder 1030. The 3rd pulse generator 1029 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (h), and outputs it to an adder 1030. Moreover, the 3rd pulse generator 1029 outputs at coincidence the location which stood the 3rd pulse to the pulse-position limited machine 1028. The pulse-position limited machine 1028 inputs the location of the 3rd pulse from the 3rd pulse generator 1029, and determines the location candidate of the 2nd pulse on the basis of the location. The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P3) of the 3rd pulse as shown in the column of the pulse number 2 of drawing 11 (h).

[0129] The pulse-position limited machine 1028 outputs the location candidate of the 2nd pulse to the 2nd pulse generator 1027. The 2nd pulse generator 1027 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 1028, and outputs it to an adder 1030. An adder 1030 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1026, 1027, and 1029, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0130] A change-over switch 1031 chooses one from a total of six kinds of noise sign vectors inputted from each adder of 1005, 1010, 1015, 1020, 1025, and 1030, and outputs the noise sign vector 1032. This selection is specified by control from the outside.

[0131] In addition, although drawing 5 (c), drawing 11 (c), (f), and (i) are prepared supposing the case where the pulse expressed in a relative position overflows a frame in drawing 5 and drawing 11 Since the range of the location candidate of the pulse expressed absolutely in a location is partial ahead of the frame, when it is not possible that the pulse expressed in a relative position overflows a frame, these parts (drawing 5 (c) etc.) can be omitted.

[0132] (Gestalt 2 of operation) Drawing 12 is the block diagram showing the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 2 of operation. The voice to digital converter shown in this drawing is equipped with the pretreatment machine 1201, the LPC analyzer 1202, the LPC quantizer 1203, the

adaptation sign book 1204, a multiplier 1205, the noise sign book 1206 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1207, an adder 1208, the LPC composition filter 1209, an adder 1210, the acoustic-sense weighting machine 1211, and the error minimization machine 1212.

[0133] In this voice to digital converter, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 1201 at every batch time amount (frame). The pretreatment machine 1201 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound signal.

[0134] The signal after pretreatment is outputted to the LPC analyzer 1202 and an adder 1210. The LPC analyzer 1202 performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 1201, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 1203. The LPC quantizer 1203 quantizes LPC inputted from the LPC analyzer 1202, outputs Quantization LPC to the LPC composition filter 1209, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.

[0135] The adaptation sign book 1204 is the buffer of the excitation vector (vector outputted from an adder 1208) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 1212, and outputs it to a multiplier 1205. A multiplier 1205 multiplies the adaptation sign vector outputted from the adaptation sign book 1204 by adaptation sign vector gain, and outputs it to an adder 1208. Adaptation sign vector gain is specified with an error minimization vessel.

[0136] The noise sign book 1206 which consists of a partial algebraic-sign book and a random sign book is a sign book with the configuration shown in drawing 14 mentioned later, and outputs either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close], or the noise sign vector of about 90% or less of rates of sparse (the measurement size of the amplitude zero to the measurement size of the whole frame comparatively) to a multiplier 1207.

[0137] A multiplier 1207 multiplies the noise sign vector outputted from the noise sign book 1206 which consists of a partial algebraic-sign book and a random sign book by noise sign vector gain, and outputs it to an adder 1208. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 1205, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1207, an adder 1208 generates an excitation vector and outputs it to the adaptation sign book 1204 and the LPC composition filter 1209.

[0138] The excitation vector outputted to the adaptation sign book 1204 is used for updating the adaptation sign book 1204, and the excitation vector outputted to the LPC composition filter 1209 is used in order to generate synthesized speech. The LPC composition filter 1209 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 1203, drives an LPC composition filter using the excitation vector outputted from the adder 1208, and outputs a composite signal to an adder 1210. the difference (error) of the input sound signal after pretreatment to which the adder 1210 was outputted from the pretreatment machine 1201, and the composite signal outputted from the LPC composition filter 1209 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 1211.

[0139] The acoustic-sense weighting machine 1211 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 1210 as an input, and outputs it to the error minimization machine 1212. The error minimization machine 1212 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 1211 as an input. For example, the square sum So that it may become min an adaptation sign vector from the adaptation sign book 1204 The value of the noise sign vector generated from the noise sign book 1206 which consists of the location and partial algebraic-sign book to cut down, and a random sign book, the adaptation sign vector gain by which it multiplies with a multiplier 1205, and the noise sign vector gain by which it multiplies with a multiplier 1207 is adjusted. Each is encoded and it outputs to a decoder side through a transmission line as sound-source parameter coded data 1214.

[0140] Drawing 13 is the block diagram showing voice decryption equipment equipped with the noise sign vector generation machine concerning the gestalt 2 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 1301, the sound-source parameter decoder 1302, the adaptation sign book 1303, a multiplier 1304, the noise sign book 1305 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1306, an adder 1307, the LPC composition filter 1308, and the after-treatment machine 1309.

[0141] In this voice decryption equipment, LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 1301 and the sound-source parameter decoder 1302 per frame through a transmission line, respectively. The LPC decoder 1301 decodes Quantization LPC and outputs it to the LPC composition filter 1308.

When using Quantization LPC with the after-treatment vessel 1309, Quantization LPC is outputted also to the after-treatment machine 1309 from the LPC decoder 1301 at coincidence. The sound-source parameter decoder 1302 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the noise sign book 1305 which consists of the adaptation sign book 1303, a multiplier 1304, and a partial algebraic-sign book and a random sign book, and a multiplier 1306.

[0142] The adaptation sign book 1303 is the buffer of the excitation vector (vector outputted from an adder 1307) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 1302, and outputs it to a multiplier 1304. A multiplier 1304 multiplies the adaptation sign vector outputted from the adaptation sign book 1303 by the adaptation sign vector gain inputted from the sound-source parameter decoder 1302, and outputs it to an adder 1307.

[0143] The noise sign book 1305 which consists of a partial algebraic-sign book and a random sign book It is a noise sign book with the configuration shown in drawing 14 , and is the same noise sign book as what was shown in 1206 of drawing 12 . Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the index inputted from the sound-source parameter decoder 1302 is close], or the noise sign vector of about 90% or less of rates of sparse is outputted to a multiplier 1306.

[0144] A multiplier 1306 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise sign vector gain inputted from the sound-source parameter decoder 1302, and outputs it to an adder 1306. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 1304, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1306, an adder 1307 generates an excitation vector and outputs it to the adaptation sign book 1303 and the LPC composition filter 1308.

[0145] The excitation vector outputted to the adaptation sign book 1303 is used when updating the adaptation sign book 1303, and the excitation vector outputted to the LPC composition filter 1308 is used in order to generate synthesized speech. The LPC composition filter 1308 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 1301, drives an LPC composition filter using the excitation vector outputted from the adder 1307, and outputs a composite signal to the after-treatment machine 1309.

[0146] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 1308 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 1309 outputs as decode voice data.

[0147] The configuration of the noise sign vector generation equipment concerning the gestalt 2 of the operation of this invention to drawing 14 is shown. The noise sign vector generation equipment shown in this drawing is equipped with the partial algebraic-sign book 1401 and the random sign book 1402 which were shown in the gestalt 1 of operation.

[0148] The partial algebraic-sign book 1401 generates the noise sign vector to which at least two pulses which consist of two or more unit pulses are close, and outputs it to a circuit changing switch 1403. The generation method of the noise sign vector of the partial algebraic-sign book 1401 is concretely shown in the gestalt 1 of operation.

[0149] The random sign book 1402 chooses one vector from the noise sign vectors which store the noise sign vector which consists of many pulse numbers, and are stored rather than the noise sign vector generated from the partial algebraic-sign book 1401, and outputs it to a change-over switch 1403.

[0150] The random sign book 1402 is [using a sign book with more independent constituting from two or more channels rather than] advantageous in respect of the amount of operations, and the amount of memory. Moreover, since a noise sign vector which two pulses are approaching is generable by the partial algebraic-sign book 1401, it can improve the engine performance to a voiceless consonant or stationary noise by storing in the random sign book 1402 the noise sign vector the pulse stands on ** etc. at the whole frame which no pulses are approaching.

[0151] Moreover, in order to lessen the amount of operations by the case where frame length is 80 samples, as for the pulse number of the noise sign vector which the random sign book 1401 stores, it is desirable to carry out to about 8-16. In this case, what is necessary is just to store the vector which consists of the pulse of each about 4-8 channels, if the random sign book 1401 is made a two-channel configuration. Moreover, it is also possible by setting the amplitude of each pulse to +1 or -1 in such a sparse vector to aim at saving of the amount of operations and the amount of memory further.

[0152] A change-over switch 1403 is control (for example, control is received from the block which performs error minimization with a target when using this noise sign vector for an encoder) from the outside. One of the noise sign vector outputted from the partial algebraic-sign book 1401 controlled by the index of the noise sign vector decoded

when using for a decryption machine, and the noise sign vectors outputted from the random sign book 1402 is chosen. It outputs as an output noise sign vector 1404 of a noise sign vector generation machine.
[0153] Here, it is desirable that they are [of the noise sign vector outputted from the random sign book 1402 and the noise sign vector outputted from the partial algebraic-sign book 1401] 1:1-2:1 [50 - 66%], i.e., random, and 34 - 50% of algebra comparatively (random: algebra).
[0154] With reference to drawing 15 , it explains below that processing of the noise sign vector generation method (the coding approach, the noise sign book retrieval approach) in the gestalt of the above-mentioned implementation flows. First, it looks for a partial algebraic-sign book in ST1501. As the detail of the concrete retrieval approach is shown in the gestalt 1 of operation, it realizes by maximizing a formula (1). The size of a partial algebraic-code book is IDXa, and the index index of the optimal candidate out of a partial algebraic-sign book (0 <=index<IDXa) is determined at this step.
[0155] Next, it looks for a random sign book in ST1502. Retrieval of a random sign book is performed using the approach which is generally performed and shines with a CELP encoder. The valuation plan shown in a formula (1) is specifically calculated to all the noise sign vectors in which it is stored by the random sign book, and the index index to the vector used as max is determined. however, since maximization of a formula (1) is performed in ST1501, only when the noise sign vector exceeding the maximum of the formula (1) determined by ST1501 exists, index determined by ST1501 is already updated on the new index index (IDXa<=index -- < (IDXa+IDXr)). When the noise sign vector exceeding the maximum of the formula (1) determined by ST1501 is not stored in the random sign book, the coded data (index index) determined by ST1501 is outputted as encoded information of a noise sign vector.
[0156] With reference to drawing 16 , the flow of processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation is explained below.
[0157] The encoded information index of the noise sign vector first transmitted and decoded from the encoder in ST1601 judges whether it is under IDXa. IDXa is the size of a partial noise sign book. This noise sign vector generation machine is generating the noise sign vector from the noise sign book which consists of the partial algebraic-sign book of size IDXa, and the random sign book of size IDXr, and, as for this noise sign book, the index is equipped with the IDXa and random - (IDXa+IDXr -1) sign book for the 0 and partial - (IDXa-1) algebraic-sign book. Therefore, with [received index] IDXa [under], a noise sign vector is generated by the partial algebraic-sign book, and with IDXa [more than] (IDXa+IDXr) (following), a noise sign vector will be generated by the random sign book. With [in this step / index] IDXa [under], it progresses to ST1602, and with IDXa [more than], it progresses to ST1604.
[0158] In ST1602, decode of a partial algebraic-sign book parameter is performed. The concrete decode approach is shown in the gestalt 1 of operation. For example, when the number of pulses is two, the location position1 of the 1st pulse and the location position2 of the 2nd pulse are decoded from Index index. Moreover, when the polar information on a pulse is also included in index, the polarity sign 1 of the 1st pulse and the polarity sign 2 of the 2nd pulse are decoded collectively. sign1 and sign2 are +1 or -1 here.
[0159] Next, in ST1603, a noise sign vector is generated from the decoded partial algebraic-sign book parameter. When the number of pulses is two, the amplitude specifically stands [a polarity] the pulse of 1 by sign1 at the location of position1, the amplitude stands [a polarity] the pulse of 1 by sign2 at the location of position2, and all the other points output the vector code [0-Num-1] set to 0 as a noise sign vector. Here, Num is frame length or noise sign vector length (sample).
[0160] On the other hand, in ST1601, when index is more than IDXa, it progresses to ST1604. In ST1604, IDXa is subtracted from index. This is for only changing index into the range of 0-IDXr-1. IDXr is the size of a random sign book here.
[0161] Next, decode of a random sign book parameter is performed in ST1605. Specifically, in the case of the random sign book of a two-channel configuration, the random sign book index indexR1 of the 1st channel and the random sign book index indexR2 of the 2nd channel are decoded from index. Moreover, when the polar information on each channel is included in index, the polarity sign 1 of the 1st channel and the polarity sign 2 of the 2nd channel are decoded collectively. sign1 and sign2 are +1 or 1.
[0162] Next, in ST1606, a noise sign vector is generated from the decoded random sign book parameter. When a random sign book is a two-channel configuration, RCB2 from 2nd channel RCB2 [indexR2] and [0-Num-1] are specifically taken out for RCB1 from 1st channel RCB1 [indexR1], and [0-Num-1], respectively, and the thing adding two vectors is outputted as a noise sign vector code [0-Num-1]. Here, Num is frame length or noise sign vector length (sample).
[0163] (Gestalt 3 of operation) Drawing 17 is the block diagram having shown the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 3 of operation. The voice to digital converter

shown in this drawing is equipped with the pretreatment machine 1701, the LPC analyzer 1702, the LPC quantizer 1703, the adaptation sign book 1704, a multiplier 1705, the noise sign book 1706 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1707, an adder 1708, the LPC composition filter 1709, an adder 1710, the acoustic-sense weighting machine 1711, the error minimization machine 1712, and the mode judging machine 1713.

[0164] In this voice to digital converter, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 1701 at every batch time amount (frame). The pretreatment machine 1701 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound signal.

[0165] The signal after pretreatment is outputted to the LPC analyzer 1702 and an adder 1710. The LPC analyzer 1702 performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 1701, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 1703. The LPC quantizer 904 quantizes LPC inputted from the LPC analyzer 903, outputs Quantization LPC to the LPC composition filter 1709 and the mode judging machine 1713, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.

[0166] The mode judging machine 1713 outputs to dynamic and the noise sign book 1716 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC. By using the dynamic description of Quantization LPC, the voice section / non-voice section is carved, and, more specifically, it performs carving between voiced / non-vocal register by using the static description of Quantization LPC. The distance (difference) of the average quantization LPC in the section judged as a dynamic description of Quantization LPC in the inter-frame amount of fluctuation and the inter-frame past to be the non-voice section and the quantization LPC in the present frame etc. can be used. Moreover, the primary reflection coefficient etc. can be used as a static description of Quantization LPC.

[0167] In addition, Quantization LPC can be more effectively used by changing into the parameter of other fields, such as LSP, a reflection coefficient, and LPC prediction remainder PAWA. Moreover, when it is possible to transmit mode information, a mode judging cannot be performed from Quantization LPC, but a more exact and fine mode judging can also be performed using various parameters which analyze input voice data and are obtained. In this case, it encodes and mode information is outputted to a decoder side through a transmission line with the LPC coded data 1714 and the sound-source parameter coded data 1715.

[0168] The adaptation sign book 1704 is the buffer of the excitation vector (vector outputted from an adder 1708) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 1712, and outputs it to a multiplier 1705. A multiplier 1705 multiplies the adaptation sign vector outputted from the adaptation sign book 1704 by adaptation sign vector gain, and outputs it to an adder 1708.

[0169] Adaptation sign vector gain is specified with an error minimization vessel. The noise sign book 1706 which consists of a partial algebraic-sign book and a random sign book As it is the noise sign book from which the ratio of a partial algebraic-sign book and a random sign book changes using the mode information inputted from the mode judging machine 1713 and is shown in <u>drawing 9</u> It has the configuration with which the number of entries of a partial algebraic-sign book and the number of entries of a random sign book are controlled by mode information accommodative (switched). Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close], or the noise sign vector of about 90% or less of rates of sparse (the measurement size of the amplitude zero to the measurement size of the whole frame comparatively) is outputted to a multiplier 1707.

[0170] A multiplier 1707 multiplies the noise sign vector outputted from the noise sign book 1706 which consists of a partial algebraic-sign book and a random sign book by noise sign vector gain, and outputs it to an adder 1708. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 1705, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1707, an adder 1708 generates an excitation vector and outputs it to the adaptation sign book 1704 and the LPC composition filter 1709.

[0171] The excitation vector outputted to the adaptation sign book 1704 is used for updating the adaptation sign book 1704, and the excitation vector outputted to the LPC composition filter 1709 is used in order to generate synthesized speech. The LPC composition filter 1709 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 1703, drives an LPC composition filter using the excitation vector outputted from the adder 1708, and outputs a composite signal to an adder 1710.

[0172] the difference (error) of the input sound signal after pretreatment to which the adder 1710 was outputted from

the pretreatment machine 1701, and the composite signal outputted from the LPC composition filter 1709 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 1711. The acoustic-sense weighting machine 1711 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 1710 as an input, and outputs it to the error minimization machine 1712.

[0173] The error minimization machine 1712 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 1711 as an input. For example, the noise sign vector generated from the noise sign book 1706 which consists of the location and partial algebraic-sign book which start an adaptation sign vector from the adaptation sign book 1704, and a random sign book so that the square sum may serve as min, The value of the adaptation sign vector gain by which it multiplies with a multiplier 1705, and the noise sign vector gain by which it multiplies with a multiplier 1707 is adjusted, each is encoded, and it outputs to a decoder side through a transmission line as sound-source parameter coded data.

[0174] Drawing 18 shows voice decryption equipment equipped with the noise sign vector generation machine concerning the gestalt 3 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 1801, the sound-source parameter decoder 1802, the adaptation sign book 1803, a multiplier 1804, the noise sign book 1805 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1806, an adder 1807, the LPC composition filter 1808, the after-treatment machine 1809, and the mode judging machine 1810.

[0175] In this voice decryption equipment, LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 1801 and the sound-source parameter decoder 1802 per frame through a transmission line, respectively. The LPC decoder 1801 decodes Quantization LPC and outputs it to the LPC composition filter 1808 and the mode judging machine 1810. When using Quantization LPC with the after-treatment vessel 1809, Quantization LPC is outputted also to the after-treatment machine 1809 from the LPC decoder 1801 at coincidence. The mode judging machine 1810 is the same configuration as the mode judging machine 1713 of drawing 17 , and outputs to dynamic, and the noise sign book 1805 and the after-treatment machine 1809 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC.

[0176] By using the dynamic description of Quantization LPC, the voice section / non-voice section is carved, and, more specifically, it performs carving between voiced / non-vocal register by using the static description of Quantization LPC. The distance (difference) of the average quantization LPC in the section judged as a dynamic description of Quantization LPC in the inter-frame amount of fluctuation and the inter-frame past to be the non-voice section and the quantization LPC in the present frame etc. can be used. Moreover, the primary reflection coefficient etc. can be used as a static description of Quantization LPC.

[0177] In addition, Quantization LPC can be used more effectively by changing into the parameter of other fields, such as LSP, a reflection coefficient, and LPC prediction remainder PAWA. Moreover, when it is possible to transmit mode information as another information, the mode information transmitted separately is decoded and decode mode information is outputted to the noise sign book 1805 and the after-treatment machine 1809.

[0178] The sound-source parameter decoder 1802 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the noise sign book 1805 which consists of the adaptation sign book 1803, a multiplier 1804, a partial algebraic-sign book, and a random sign book, and a multiplier 1806.

[0179] The adaptation sign book 1803 is the buffer of the excitation vector (vector outputted from an adder 1807) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 1802, and outputs it to a multiplier 1804. A multiplier 1804 multiplies the adaptation sign vector outputted from the adaptation sign book 1803 by the adaptation sign vector gain inputted from the sound-source parameter decoder 1802, and outputs it to an adder 1807.

[0180] The noise sign book 1807 which consists of a partial algebraic-sign book and a random sign book It is a noise sign book with the configuration shown in drawing 9 , and is the same noise sign book as what was shown in 1706 of drawing 17 . Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the index inputted from the sound-source parameter decoder 1802 is close], or the noise sign vector of about 90% or less of rates of sparse is outputted to a multiplier 1806.

[0181] A multiplier 1806 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise sign vector gain inputted from the sound-source parameter decoder 1802, and outputs it to an adder 1806. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 1804, and the noise sign vector after the noise sign vector gain multiplication outputted from the

multiplier 1806, an adder 1807 generates an excitation vector and outputs it to the adaptation sign book 1803 and the LPC composition filter 1808.
[0182] The excitation vector outputted to the adaptation sign book 1803 is used for updating the adaptation sign book 1803, and the excitation vector outputted to the LPC composition filter 1808 is used in order to generate synthesized speech. The LPC composition filter 1808 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 1801, drives an LPC composition filter using the excitation vector outputted from the adder 1807, and outputs a composite signal to the after-treatment machine 1809.
[0183] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 1808 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 1809 outputs as decode voice data 1810. Such after treatment is performed accommodative using the mode information inputted from the mode judging machine 1808. That is, the after treatment for which it was suitable for every mode is changed and applied, or the strength of after treatment is changed accommodative.
[0184] Drawing 19 is the block diagram showing the configuration of the noise sign vector generation equipment concerning the gestalt 3 of operation of this invention. The noise sign vector generation machine shown in this drawing is equipped with the pulse-position limited machine controller 1901, the partial algebraic-sign book 1902, the number controller 1903 of random sign book entries, and the random sign book 1904.
[0185] The pulse-position limited machine controller 1901 outputs the control signal of a pulse-position limited machine to the partial algebraic-sign book 1902 according to the mode information inputted from the outside. Size of a partial algebraic-sign book is made small by the thing which this control makes fluctuate size of a partial algebraic-sign book (responding to the mode) and for which it carries out for accumulating, and limitation is strengthened case [whose mode is / like silent / stationary noise mode] (the number of candidates of the pulse position is lessened) (instead, it controls by the number controller 1903 of random sign book entries so that the size of the random sign book 1904 becomes large).
[0186] By doing in this way, if the noise sign vector which consists of several pulses, such as the silent section and the stationary noise section, is used, it will become possible to aim at an engine-performance improvement to a signal with which subjective quality deteriorates. The pulse-position limited machine is built into the partial algebraic-sign book 1902, and the concrete actuation is shown in the gestalt 1 of operation.
[0187] The partial algebraic-sign book 1902 is a partial algebraic-sign book by which actuation of the pulse-position limited machine built into the interior is controlled by the control signal inputted from the pulse-position limited machine controller 1901, and sign book size fluctuates it by the limited degree of a pulse-position candidate with a pulse-position limited machine. Concrete actuation of a partial algebraic-sign book is shown in the gestalt 1 of operation. The noise sign vector generated from this sign book is outputted to a change-over switch 1905.
[0188] The number controller 1903 of random sign book entries performs control which fluctuates the size of the random sign book 1904 according to the mode information inputted from the outside. This control is performed by control of the pulse-position limited machine controller 1901 being interlocked with. That is, when the number controller 1903 of random sign book entries decreases the size of the random sign book 1904 when the size of the partial algebraic-sign book 1902 is made to increase with the pulse-position limited machine controller 1901, and decreasing the size of the partial algebraic-sign book 1902 with the pulse-position limited machine controller 1901, the number controller 1903 of random sign book entries performs control to which the size of the random sign book 1904 is made to increase. And the total number of entries (total sign book size in this noise sign vector generation machine) which set the partial algebraic-sign book 1902 and the random sign book 1904 is always maintained at a fixed value.
[0189] The random sign book 1904 generates a noise sign vector using the random sign book of the size which inputted the control signal from the number controller 1903 of random sign book entries, and was specified, and outputs it to a change-over switch 1905. It is more effective from the field of the amount of memory to use it as a random sign book of two or more sizes by consisting of only random sign books of one kind to share of existing size which was defined, and using this partially, although the random sign book 1904 may consist of random sign books of size by which plurality differs here.
[0190] Moreover, although a sign book independent one channel is sufficient as the random sign book 1904, it is more advantageous from the field of the amount of operations, or the amount of memory to use the sign book which consists of two or more two or more channels.
[0191] a change-over switch 1905 -- the control (the control signal from the block which minimizes an error with a target vector when using this noise sign vector generation machine for an encoder --) from the outside Using the

parameter information on the noise sign book decoded when using for a decryption machine etc. One of the noise sign vectors outputted from the partial algebraic-sign book 1902 or the random sign book 1904 is chosen, and it outputs as an output noise sign vector 1906 of this noise sign vector generation machine.

[0192] Here, it is [in / comparatively (random: algebra) / voiced mode] desirable that they are [of the noise sign vector outputted from the random sign book 1904 and the noise sign vector outputted from the partial algebraic-sign book 1902] 0:1-1:2 [0 - 34%], i.e., random, and 66 - 100% of algebra. Moreover, as for the above-mentioned rate (random: algebra), in non-voiced mode, it is desirable that they are 2:1-4:1 [66 - 80%], i.e., random, and 20 - 34% of algebra.

[0193] With reference to drawing 20 , it explains below that processing of the noise sign vector generation method (the coding approach) in the gestalt of the above-mentioned implementation flows.

[0194] First, in ST2001, size of a partial algebraic-sign book and a random sign book is set up based on the mode information inputted separately. At this time, a setup of the size of a partial algebraic-sign book is performed by fluctuating the number of location candidates of the pulse which is shown in the gestalt 1 of operation and by which a relative-position expression is carried out.

[0195] This change in a pulse by which a relative-position expression is carried out can be performed mechanically, and it is made to decrease by reducing from the part which a relative position leaves. When relative positions are {1, 3, 5, 7}, more specifically, the number of location candidates is reduced like {{1, 3, 5}, {1, 3}, 1}. Conversely, like {1} to {1, 3}, and {1, 3, 5}, when increasing, it increases.

[0196] Moreover, a setup of the size of a partial algebraic-sign book and a random sign book is performed so that total of the size of a partial algebraic-sign book and a random sign book may become constant value. In the mode which corresponds to the voiced (stationary) section, the size (ratio) of a partial algebraic-sign book is more specifically large, and size of both the sign book is set up so that the size (ratio) of a random sign book may become large in the mode which corresponds to the silent section or the noise section.

[0197] the mode information and IDXa which mode inputted in this block -- the size (the number of noise sign vector entries) of a partial algebraic-sign book, and IDXr -- random sign book size (the number of noise sign vector entries) -- it is -- IDXa+IDXr= constant value -- it comes out. Moreover, a setup of the number of entries of a random sign book is realizable by setting up the range of the random sign book referred to, for example. For example, in control which switches and uses the size of the random sign book of two channels by 128x128=16384 and 64x64=4096, it is easily realizable by switching the range of an index which it has the random (indexes 0-127) sign book which stores the vector of 128 kinds of each channel, respectively, and is searched for it by two kinds, 0-127, and 0-63.

[0198] In addition, as for the vector space where the vector of indexes 0-127 exists in this case, and the vector space where the vector of indexes 0-63 exists, it is desirable that it is in agreement as much as possible. If the vector of indexes 64-127 cannot be expressed at all, namely, the vector space of indexes 0-63 completely differs from the vector space of indexes 64-127 by the vector of indexes 0-63 Since modification of the above random sign book sizes may degrade the coding engine performance of a random sign book greatly, it needs to create a random sign book in consideration of such a thing.

[0199] In addition, inevitably, when keeping constant total of the number of entries of a partial algebraic-sign book and a random sign book, since the method (put together) of sizing of both the sign book is limited to some kinds, it serves as ** [switch / control of sizing / a setup of these some kinds]. In Book ST, the partial algebraic-sign book size IDXa and the random sign book size IDXr are set up from the inputted mode information mode.

[0200] Next, in ST2002, the noise sign vector which makes an error with a target vector the smallest is chosen from a partial algebraic-sign book (size IDXa) and a random sign book (IDXr), and it asks for the index. It is determined that it will become the range of - (IDXa+IDXr -1) if a noise sign vector is chosen for example, from a partial algebraic-sign book and Index index will be chosen from a 0 - (IDXa-1) random sign book (IDXa-1).

[0201] Next, in ST2003, the called-for index index is outputted as coded data. index is encoded by the form further outputted to a transmission line if needed.

[0202] With reference to drawing 21 , it explains below that processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation flows.

[0203] First, in ST2101, size of a partial algebraic-sign book and a random sign book is set up based on the mode information mode decoded separately. The approach of a concrete setup is as the above-mentioned explained with reference to drawing 20 . The size IDXa of a partial algebraic-sign book and the size IDXr of a random sign book are set up from the mode information mode.

[0204] Next, in ST2102, a noise sign vector is decoded using a partial algebraic-sign book or a random sign book. it determines using which sign book it decodes by the value of the index index of the noise sign vector decoded

separately -- having -- the case of 0 <=index<IDXa -- IDXa<=index from a partial algebraic-sign book -- < (IDXa+IDXr) -- a case is decoded from a random sign book. With the gestalt 3 of operation, with reference to drawing 16 , as it explained, specifically, it decodes.
[0205] In addition, if the above indexes are given, a different index to the entry of the noise sign vector shared in the different mode will be given. Since it becomes easy to be influenced of [when (that is, it becoming an index which is different when the modes' differ also by the noise sign vector which has the completely same configuration), and a transmission-line error arise] If the same index is given to the entry of the noise sign vector shared in the mode which is different in order to avoid this, said noise sign vector generation equipment with error resistance is realizable. An example is shown in drawing 22 and drawing 23 .
[0206] Drawing 22 is the example which combined the noise sign book size 32, 11 or more samples of frame (factice) length, and the partial algebraic-sign book and the two-channel random CB of a pulse number 2, and the vector which a pulse approaches in a frame (factice) tail is a type which is not taken into consideration.
[0207] On the other hand, drawing 23 is the example which combined the noise sign book size 16, frame (factice) length 8 sample, and the partial algebraic-sign book and the two-channel random CB of a pulse number 2, and is a type with which the vector which a pulse approaches in a frame (factice) tail is also taken into consideration.
[0208] In drawing 22 and both drawings of drawing 23 , the column of eye one train shows the noise sign book index [as opposed to / in the column of eye three trains / channel / of 1st pulse or random sign book / 1st / column / of eye two trains / each combination for the 2nd channel, the 2nd pulse or a random sign book,], respectively.
[0209] (a) of both drawings moreover, the case where the ratio of a random sign book is low (the number of entries is) and where the ratio of a partial algebraic-sign book is high (there are many entries) (b) shows the case where the ratio of a random sign book is high (the number of entries is) and where the ratio of a partial algebraic-sign book is low (there are few entries), respectively, and only the noise sign vector corresponding to the index with which it added shading in the slash differs from (a) by (b).
[0210] In drawing 22 and drawing 23 , the figure to which P1 and P2 gave the 1st of a random sign book and the 2nd channel to Ra, and Rb gave the 1st and 2nd pulse positions for the pulse position [in / in the figure (except for an index) of front Naka / a partial algebraic-sign book] to Ra and Rb shows the number of the random sign vector stored in both channels, respectively. Respectively corresponding to [if the partial algebraic-sign book of drawing 5 is made to correspond / the indexes 0-5 of drawing 23 , and the indexes 0-7 of drawing 22 / drawing 5 / (a) / the indexes 10-11 of drawing 23 / drawing 5 (c)] (in drawing 22 , there is no part corresponding to drawing 5 (c)) corresponding to drawing 5 (b) in the indexes 6-9 of drawing 23 , and the indexes 8-15 of drawing 23 .
[0211] When decoding, for example, as 11 or less index of drawing 23 (a) explained the index to which it added shading with the slash from having stood in a line regularly within limited limits in drawing 22 and both drawings of drawing 23 using drawing 9 Decode (IDX 1= 6, IDX 2= 10), and in drawing 23 (b), an index is 11 or less, and the decode same only when the number is even as the case of drawing 23 (a) is performed. When the number is odd, it is possible to judge the quotient which divided the index by 2 the index corresponding to a random sign book, and to decode the vector number of each channel of a random sign book.
[0212] The same thing can be said also in drawing 22 and the vector number of an index and a random sign book can be made to correspond within the limits of the defined index regularly. Moreover, also when encoding, it is possible to carry out another treatment only of the part of the index from which it thinks the same way and a random sign book and a partial algebraic-sign book change by change in the mode, and to encode it.
[0213] thus, since only the noise sign vector corresponding to a part of indexes can be influenced by the mode of a change by carrying out, it is also possible to suppress the effect of [when the mode is mistaken with a transmission-line error] to the minimum. In such a case, although how to attach Index index compared with the case where it explains with reference to the above-mentioned flow Fig. (drawing 6 , 9, 15, 16, 20, 21) changes, the fundamental sign book retrieval approach is the same.
[0214] Thus, the coding engine performance to non-vocal sound voice or a background noise is improvable, suppressing quality degradation at the time of a mode judging error by changing the use rate of an algebraic-sign book and a random sign book by mode judging.
[0215] (Gestalt 4 of operation) With the gestalt of this operation, PAWA of an excitation signal is computed, when voice mode is noise mode, average power is computed from PAWA of an excitation signal, and the case where the number of predetermined pulse-position candidates is made to fluctuate based on this average power is explained.
[0216] Drawing 24 is the block diagram showing the configuration of the voice to digital converter concerning the gestalt 4 of operation of this invention. The voice to digital converter shown in drawing 24 has the almost same configuration as the voice to digital converter shown in drawing 17 . In the configuration shown in drawing 24 , it has

the present PAWA calculation machine 2402 which computes present PAWA from an excitation signal, and the average power calculation machine 2401 between noise divisions which computes average power from PAWA of an excitation signal based on present PAWA from the mode judging information from the mode judging machine 1713, and the present PAWA calculation machine 2402 when voice mode is noise mode.

[0217] The mode judging machine 1713 outputs to dynamic and the noise sign book 1716 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC, as the gestalt 3 of operation explained. Moreover, the mode information from the mode judging machine 1713 is sent to the average power calculation machine 2401 between noise divisions.

[0218] On the other hand, PAWA of an excitation signal is computed with the present PAWA calculation vessel 2402. Thus, PAWA of an excitation signal is supervised. This present PAWA calculation result is sent to the average power calculation machine 2401 between noise divisions.

[0219] With the average power calculation vessel 2401 between noise divisions, the average power of the noise section is computed based on the calculation result and mode judging result from the present PAWA calculation machine 2402. The calculation result of present PAWA is serially inputted into the average power calculation machine 2401 between noise divisions from the present PAWA calculation machine 2402. And with the average power calculation vessel 2401 between noise divisions, when the information that it is the noise section is inputted from the mode judging machine 1713, the average power of the noise section is computed using the calculation result of inputted present PAWA.

[0220] The calculation result of this average power is sent to a good transformation subalgebra sign book / random sign book 1706. Based on the calculation result of average power, the use ratio of an algebraic-sign book and a random sign book is controlled by the good transformation subalgebra sign book / random sign book 1706. About the approach of this control, it is the same as the gestalt 3 of operation.

[0221] In addition, with the average power calculation vessel 2401 between noise divisions, the comparison with present PAWA serially inputted as the computed average power between noise divisions is performed. And since it is thought that a problem is in an average power value when the average power of the noise section is larger than present PAWA, the average power of the noise section is updated to present PAWA. The use ratio of an algebraic-sign book and a random sign book can be controlled with a thereby more sufficient precision.

[0222] Moreover, drawing 25 is the block diagram showing the configuration of the voice decryption equipment concerning the gestalt 4 of operation of this invention. The voice decryption equipment shown in drawing 25 has the almost same configuration as the voice decryption equipment shown in drawing 18 . In the configuration shown in drawing 25 , it has the present PAWA calculation machine 2502 which computes present PAWA from an excitation signal, and the average power calculation machine 2501 between noise divisions which computes average power from PAWA of an excitation signal based on present PAWA from the mode judging information from the mode judging machine 1810, and the present PAWA calculation machine 2502 when voice mode is noise mode.

[0223] The mode judging machine 1810 outputs to dynamic, and the noise sign book 1805 and the after-treatment machine 1809 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC, as the gestalt 3 of operation explained. Moreover, the mode information from the mode judging machine 1810 is sent to the average power calculation machine 2501 between noise divisions.

[0224] On the other hand, PAWA of an excitation signal is computed with the present PAWA calculation vessel 2502. Thus, PAWA of an excitation signal is supervised. This present PAWA calculation result is sent to the average power calculation machine 2501 between noise divisions.

[0225] With the average power calculation vessel 2501 between noise divisions, the average power of the noise section is computed based on the calculation result and mode judging result from the present PAWA calculation machine 2502. The calculation result of present PAWA is serially inputted into the average power calculation machine 2501 between noise divisions from the present PAWA calculation machine 2502. And with the average power calculation vessel 2501 between noise divisions, when the information that it is the noise section is inputted from the mode judging machine 1810, the average power of the noise section is computed using the calculation result of inputted present PAWA.

[0226] The calculation result of this average power is sent to a good transformation subalgebra sign book / random sign book 1805. Based on the calculation result of average power, the use ratio of an algebraic-sign book and a random sign book is controlled by the good transformation subalgebra sign book / random sign book 1805. About the approach of this control, it is the same as the gestalt 3 of operation.

[0227] In addition, with the average power calculation vessel 2501 between noise divisions, the comparison with

present PAWA serially inputted as the computed average power between noise divisions is performed. And since it is thought that a problem is in an average power value when the average power of the noise section is larger than present PAWA, the average power of the noise section is updated to present PAWA. The use ratio of an algebraic-sign book and a random sign book can be controlled with a thereby more sufficient precision.

[0228] Here, when [of the noise sign vector outputted from a random sign book, and the noise sign vector outputted from a partial algebraic-sign book] the level of the noise section is large comparatively (random: algebra), in voiced mode, it is [2:1, i.e., random about 66%,] desirable that it is about 34% of algebra. Moreover, as for the above-mentioned rate (random: algebra), in non-voiced mode, it is desirable that it is about 2% of algebra random about 98%.

[0229] Thus, the coding engine performance to non-vocal sound voice or a background noise can be raised, suppressing quality degradation at the time of a mode judging error by supervising the noise section and changing the use rate of an algebraic-sign book and a random sign book by mode judging.

[0230] In addition <A HREF="/Tokujitu/tjitemdrw.ipdl?N0000=237&N0500=1 E_N/;>=<==?68///&N0001=545&N0552=9&N 0553= 000027" TARGET = Although the case where present PAWA is computed from an excitation signal is explained, you may make it compute present PAWA in this invention in "tjitemdrw"> drawing 24 and drawing 25 using PAWA of the composite signal after LPC composition.

[0231] The above-mentioned voice to digital converter and/or voice decryption equipment can be used for the communication terminal or base station equipments of migration communication equipment, such as a cellular phone. [, such as a migration machine,] In addition, the medium which transmits information is possible not only for an electric wave as shown in the gestalt of this operation but using a lightwave signal etc., and possible also for using the transmission line of a cable further.

[0232] In addition, it records on record media, such as a magnetic disk, a magneto-optic disk, and a ROM cartridge, as software, and voice coding / decryption equipment shown in the gestalt of the above-mentioned implementation can also be realized. By using the record medium, if the personal computer which uses such a record medium realizes a voice to digital converter / decryption equipment, and a sending set/receiving set, it can **.

[0233] (Gestalt 5 of operation) The gestalt of this operation explains the case where the algebraic-sign book the number of sound-source pulses is [book] three is used as a noise sign book. Here, the case where 16 bits per subframe are assigned to a noise sign book is explained. In addition, in the gestalt of this operation, an algebraic-sign book and the random sign book which has arranged the sound-source pulse to the whole subframe at homogeneity are used together.

[0234] In this case, in order to use a random sign book together, without changing the number of bits of the whole noise sign book, reduction of the size of an algebraic-sign book is needed. If algebraic-sign book size is reduced simply, the retrieval location candidate of each pulse must be reduced and wide range retrieval will become difficult. Then, algebraic-sign book size is reduced, with the retrieval range of a sound-source pulse maintained.

[0235] Specifically paying attention to the configuration of the sound-source vector generated from an algebraic-sign book, the sound-source vector which has the configuration where operating frequency is low reduces the size of an algebraic-sign book by adding a limit so that it may not be generated from an algebraic-sign book. As characteristic quantity which shows the configuration of a sound-source vector, the relative location of each sound-source pulse is used. That is, as shown in drawing 26 , the spacing B of the pulse 2601 of the head of a sound-source vector, the spacing A of the 2nd pulse 2602 and the 2nd pulse 2602 which were constituted by three sound-source pulses 2601-2603, and the 3rd pulse 2603 is used. A vector with low operating frequency is determined based on such characteristic quantity, the size of an algebraic-sign book is reduced, and a random sign book is used together. Thus, since the algebraic-sign book is being partially used for the algebraic-sign book which reduced size, suppose that it is called a partial algebraic-sign book.

[0236] In order to examine the construction of a partial algebraic-sign book, the vector configuration where operating frequency was low was investigated using the spacing A shown in drawing 26 , and spacing B. Since two or more sound-source vectors which have spacing A and spacing B existed, they were normalized by the number of the combination which may be generated from a partial algebraic-sign book. Moreover, since it was possible that inclinations differ in the voiced section and the non-voiced section, the voiced section and non-**** were classified using the primary reflection coefficient etc., and use frequency distribution was investigated about each.

[0237] Investigation showed that the operating gradient of a vector with either [at least] spacing A or the spacing B narrow in the voice section is high, and becoming frequency distribution uniform on the whole compared with the voiced section in the non-voiced section. The partial algebraic-sign book consisted of these results of an investigation by adding a limit so that only a vector with at least 1 set of narrow sound-source pulse separations may be generated.

[0238] The following two approaches are mentioned as an approach of generating only a vector with at least 1 set of narrow sound-source pulse separations.

(Approach 1) In a partial algebraic-sign book, all retrieval is performed, it judges whether the sound-source pulse separation under current retrieval are narrower than a predetermined distance in a retrieval loop formation, and only a narrow thing is made applicable to retrieval.
(Approach 2) It searches only for combination from which the difference of the index of each sound-source pulse is set to (K) predetermined within the limits in a partial algebraic-sign book. It specifically classifies into three kinds of patterns as shown in drawing 27 (a) - (c) (when drawing 27 (a):3 pulse is near and two pulses after drawing 27 (c): are [two pulses in front of drawing 27 (b): are near and] near), and looks for a partial algebraic-sign book. however, drawing 27 (a) - (c) shows only the case where it stands in a line in order of pulses 2601-2603, and it can think by it as sequence that these three pulses are located in a line in fact -- combination is altogether taken into consideration.
[0239] Although the limit by the distance of pulse separation is strictly possible when an approach 1 is used, conditional branching is needed within a retrieval loop formation each time. On the other hand, by the approach 2, in the case of an uneven retrieval location candidate, although it stops being the limit by strict pulse-separation distance, it becomes possible to search only for the required part of an algebraic-sign book regularly, and conditional branching within a retrieval loop formation becomes unnecessary.
[0240] Thus, by setting a sound-source pulse as three pulses, and constituting a partial algebraic-sign book, the high partial algebraic-sign book of fundamentality ability is realizable.
[0241] Next, the random sign book used together with the above-mentioned partial algebraic-sign book is explained. In order to improve expression nature of a vector which PAWA is distributing to the whole subframe, this random sign book is constituted so that a sound-source pulse may be arranged if possible at ** etc. at the whole subframe. By this random sign book, pulse amplitude was set to **1, and the pulse position is restricted so that a sound-source pulse may not lap between each channel (ch). Moreover, a random number generates the location and amplitude (polarity) of a sound-source pulse. A sound-source pulse number shows the random sign book of 2ch configurations in total to drawing 28 by eight.
[0242] This random sign book performs a setup of the number of channels, and a pulse number, sets up the arrangement range of each pulse, and creates it by making the location / polar decision of each pulse. after performing a setup of the number of channels, and a pulse number first in the creation approach of this random sign book -- the arrangement range of each pulse -- setting up . That is, the range length (N_Range [i], [j]) by which each pulse is arranged is set up. As this setup is shown in drawing 29 , it is performed.
[0243] First, subframe length is divided with a pulse number (one channel), N_Range0 is calculated, and a remainder is saved as N_Rest (ST2901). Subsequently, N_Range0 is divided with the number of channels, and N_Range [i] and [j] are set up (ST2902). Here, i shows a channel number and j shows a pulse number. When N_Range0 cannot divide among the number (N_ch) of channels at this time, that remainder is assigned in an order from the younger one of a channel number (ST2902).
[0244] Subsequently, N_Rest is assigned in an order from N_Range [N_ch -1] of the pulse arranged in the subframe tail end, and [N_Pulse -1] (ST2903). This completes a setup of N_Range [i] and [j].
[0245] In a setup of the arrangement range of each pulse, the start point (S_Range [i], [j]) of N_Range [i] and [j] is set up. That is, when N_Range [i] and [j] have been arranged sequentially from a subframe head, it asks for each head location. As a setup of this start point is shown in drawing 30 , it is performed. First, S_Range [i] of the head pulse of each channel and [0] are determined. In this case, it carries out in an order from the one where a pulse number is younger (ST3001). Subsequently, remaining S_Range [i] and [0] are determined similarly (ST3002). Thus, a setup of S_Range [i] and [j] is completed.
[0246] As mentioned above, after setting up the arrangement range of each pulse, the location / polar decision of each pulse is made. As the location/polarity of each of this pulse are shown in drawing 31 , it is performed. First, the loop counter of a channel is reset (ST3101). Subsequently, it judges whether a loop counter i is smaller than N_ch (ST3102). A counter and a threshold will be reset if a loop counter i is smaller than N_ch (ST3103). That is, the number (counter) of the determined random sign vectors, the number (counter_r) which repeated generation of a random sign vector, and the pulse number (thresh) which allows locations to differ are reset. On the other hand, if a loop counter i is not smaller than N_ch, creation of a random sign book is ended.
[0247] Subsequently, it judges whether the number (counter_r) which repeated generation of a random sign vector is maximum MAX_r (ST3104). If counter_r is not MAX_r, generation of a code vector, and the pulse position and the polar generation by the random number will be performed (ST3106), if counter_r is MAX_r, a threshold (thresh) will be incremented and a repeat counter (counter_r) will be reset (ST3105). And generation of a code vector, and the pulse position and the polar generation by the random number are performed (ST3106). In addition, in the pulse position and the polar generation by the random number, rand ( ) expresses an integer random-number generating function.

[0248] Subsequently, a code vector is checked after generating the pulse position and a polarity (ST3107). Here, the generated code vector is compared with all the code vectors already registered into the random sign book, and it is confirmed whether the code vector with which the pulse position laps exists. And the pulse number with which the location has lapped for every code vector is counted.
[0249] Subsequently, it judges whether there is any code vector to which the pulse number to which a location laps in a random sign book exceeded the threshold (ST3108). If there is a code vector to which the pulse number with which a location laps exceeded the threshold, the counter (counter_r) to repeat will be incremented (ST3109) and it will progress to ST3104 after that. If there is no code vector to which the pulse number with which a location laps exceeded the threshold on the other hand, the code vector will be registered into a random sign book (ST3110). That is, the code vector generated with the random number is stored in a random sign book, and a counter (counter) is incremented.
[0250] Subsequently, a counter (counter) judges whether it is more than the size of a random sign book (ST3111). If it is more than the size of the random sign book which a counter (counter) creates, the loop counter of a channel will be incremented (ST3112) and it will progress to ST3102. If a counter (counter) is not more than the size of a random sign book, it will progress to ST3104.
[0251] In creation of this random sign book, with a random number, the pulse position of a code vector and a polarity are determined, and it checks so that a pulse and a location may not lap. [finishing / decision / already] Thus, that with which a location does not lap at all is generated in the beginning, and the pulse number with which a sequential location laps is made to increase.
[0252] Moreover, in creation of a random sign book, when the whole subframe is divided into ** etc. and cannot be divided [**] completely, the range of ch1 is made large from ch2, and the range is made large from the direction of a subframe tail. For example, it explains using drawing 32 . The figure shows the arrangement range (N_Range [i], [j]) and start point (S_Range [i], [j]) of each pulse (pulse number j), and it is indicated that drawing 32 goes to the tail of a subframe toward the bottom from a top. In drawing 32 (a), since it is four pulses, 80 samples of the whole subframe can be divided into ** etc. In drawing 32 (b), since it is six pulses, 80 samples of the whole subframe cannot be divided into ** etc. In this case, ch1 (7) is made larger than ch2 (6), and, moreover, the subframe tail (ch1:8, ch2:7) is made large. The range of ch1 is made larger than ch2 because it assumes making [more] the number of code vectors of ch1 (sign book size) than the number of code vectors of ch2. in addition, the value of N_Range [i] of ch1 and ch2, and [j] -- ******** -- it is made like and assigning an odd part to each channel ** etc. in the second half of a subframe is also considered.
[0253] Thus, by creating a random sign book, the random sign book by which a sound-source pulse is distributed over the whole subframe can be created efficiently. Moreover, since the lapping sound-source pulse increases so that it becomes in the second half of a sign book, when making sign book size small, a desirable sign book can be created by reducing from the section in the second half.
[0254] Next, the case where a mode change is applied is explained in concomitant use of a partial algebraic-sign book and a random sign book. In this case, according to the shape of a sound-source pulse form, the block division of the partial algebraic-sign book is carried out, it reduces gradually corresponding to that block, and it is made to increase a random sign book gradually (accommodative).
[0255] Drawing 33 is drawing showing the condition of having carried out the block division of the partial algebraic-sign book. The block division is performed corresponding to the shape of a sound-source pulse form. This block is determined by the spacing A and B (correctly difference of an index) between the pulses of the sound-source pulse shown in drawing 34 (a). That is, Blocks X-Z support the field shown in drawing 34 (b).
[0256] Thus, by carrying out a block division and reducing the size of a partial algebraic-sign book, size is easily controllable. What is necessary is just to specifically turn OFF the retrieval loop formation of the corresponding block.
[0257] Thus, while dividing a partial algebraic-sign book into a block, the phase division of the random sign book is carried out. Here, as shown in drawing 35 (a), a phase division is carried out by ch1 and ch2 at a three-stage. The 1st step is set to a and b, the 2nd step is set to c and d, and, specifically, the 3rd step is set to e and f. Partial algebraic-sign books are reduced per block using these, and only the part increases a random sign book gradually, and enlarges the rate of a random sign book. The mode is determined corresponding to reduction of partial algebraic-sign books, and the increment in a random sign book. Specifically, the mode shown in drawing 33 (a) - (c) is determined. In addition, when it is instantiation and carries out mode setting about this mode number more coarsely than drawing 33 , the 2 mode may be used, and when carrying out mode setting more finely than drawing 33 , the four or more modes may be used.
[0258] The random sign book used for every mode of this is explained using drawing 33 and drawing 35 . (a) and the largest mode are set to (c) and the middle mode is set to (b) for the mode in which random sign book size is the smallest. Mode (a) -> (b) When making it change with -> (c), as for the random sign book of ch1, in drawing 35 , size

of the random sign book [a->(a+c)->(a+c+e) and] of ch2 increases like b->(b+d)->(b+d+f). Since the same index is given in every mode to a common code vector in each mode at this time, how to assign the following indexes is used.

[0259] First, the index of the vector generated by axb is assigned. Then, the index of the vector generated by cxb and (a+c) xd is assigned. The index of the vector finally generated by x(a+c+e) f and ex (b+d) is assigned. An example of this assigning method is shown in drawing 33 .

[0260] Therefore, when using together a partial number-of-stages sign book and a random sign book and a partial algebraic-sign book consists of blocks X, Y, and Z, a random sign book serves as a part shown in drawing 35 (b) of a random sign book, as shown in drawing 33 (a). Moreover, when a partial algebraic-sign book consists of Blocks X and Y, a random sign book serves as a part shown in drawing 35 [of a random sign book] (b) - (d), as shown in drawing 33 (b). Moreover, when a partial algebraic-sign book consists of block X, a random sign book serves as a part shown in drawing 35 [of a random sign book] (b) - (f), as shown in drawing 33 (c).

[0261] This mode change is performed according to the mode information which is a control signal from a mode judging machine. This mode information may decode the various information (an LPC parameter, gain parameter, etc.) transmitted from an encoder side, and may generate it according to that information, and the mode information transmitted from the encoder side may be used.

[0262] Thus, the size of a partial algebraic-sign book and a random sign book is easily controllable by reducing partial algebraic-sign books per block, and increasing a random sign book gradually. Furthermore, since a share code vector index can be made the same also in the different mode, the effect of a mode error can be suppressed.

[0263] Here, the example of the percentage of the partial algebraic-sign book in each mode and a random sign book is shown by making into an example the case where the mode consists of three kinds of modes of voiced / silent / stationary noise. Although this rate of optimum ratio may change by bit allocation In the example of a 16-bit random sign book voiced mode -- (-- partial algebraic-sign book: -- random sign book = -- about 50% : about 50%) -- silent mode -- (-- **= -- about 10% : about 90%) and stationary noise mode -- (-- **= -- if there are very few about 10% : about 90% and mode errors -- **= -- the ratio of a random sign book may be gathered to about 0% : about 100% --) -- ** -- it is desirable that it is the said ratio. In addition, also when after treatment which raises the subjectivity quality of a stationary noise signal by the decoder side is added, and it becomes unnecessary to make high especially the ratio of the random sign book in stationary noise mode, it is.

[0264] (Gestalt 6 of operation) In the gestalt of this operation, the case where switch the noise nature of a diffusion pattern by the height of noy ZUPAWA (average power in the past noise mode section), or the sampled value of 1 sample eye of a diffusion pattern is operated by the height of noy ZUPAWA is explained.

[0265] Drawing 36 is the block diagram showing the configuration of the voice to digital converter concerning the gestalt 6 of operation of this invention, and drawing 37 is the block diagram showing the configuration of the voice decryption equipment concerning the gestalt 6 of operation of this invention. In drawing 36 , about the same part as drawing 24 , the same sign as drawing 24 is attached and detailed explanation is omitted. Moreover, in drawing 37 , about the same part as drawing 25 , the same sign as drawing 25 is attached and detailed explanation is omitted.

[0266] In the voice to digital converter shown in drawing 36 , it has the good transformation partial algebraic-sign book / random sign book 3601, and has the pulse diffuser 3602 which diffuses the pulse of the sound-source vector outputted from this good transformation partial algebraic-sign book / random sign book 3601. Diffusion of the pulse of this sound-source vector is performed according to the diffusion pattern generated in the diffusion pattern generation machine 3603. The height of the average power between noise divisions called for with the average power calculation vessel 2401 between noise divisions and the mode information from the mode judging machine 1713 determine this diffusion pattern.

[0267] In the voice decryption equipment shown in drawing 37 , corresponding to the voice to digital converter shown in drawing 36 , it has the good transformation partial algebraic-sign book / random sign book 3701, and has the pulse diffuser 3702 which diffuses the pulse of the sound-source vector outputted from this good transformation partial algebraic-sign book / random sign book 3701. Diffusion of the pulse of this sound-source vector is performed according to the diffusion pattern generated in the diffusion pattern generation machine 3703. The height of the average power between noise divisions called for with the average power calculation vessel 2501 between noise divisions and the mode information from the mode judging machine 1810 determine this diffusion pattern.

[0268] With the diffusion pattern generation vessels 3603 and 3703 in the voice decryption equipment shown in the voice to digital converter shown in drawing 36 , and drawing 37 , as it is shown in drawing 38 and drawing 39 , a diffusion pattern is generated.

[0269] First, in a voice to digital converter, the average power between noise divisions is computed with the average power calculation vessel 2401 between noise divisions using PAWA of the frame (factice) judged in the past to be the

noise section. Past noise section PAWA is serially updated using PAWA outputted with the present PAWA calculation vessel 2402. The average power of the noise section computed here is outputted to the diffusion pattern generation machine 3603. With the diffusion pattern generation vessel 3603, the noise nature of a diffusion pattern is switched based on the average power of the noise section. That is, as shown in drawing 38 , with the diffusion pattern generation vessel 3603, two or more noise nature is set up corresponding to the height of the average power of the noise section, and noise nature is chosen according to the height of average power. When the average power of the noise section is large, the noise nature of a diffusion pattern chooses a high (strong) thing, and when the average power of the noise section is small, specifically, the noise nature of a diffusion pattern chooses a low (weak) thing.
[0270] Moreover, you may make it switch the noise nature of a diffusion pattern in the noise section and the voice section. In addition, the voice section may be further divided between the voiced section and non-vocal register. In this case, in the noise section, the noise nature of a change of a diffusion pattern is high, and it is performed in the voice section so that the noise nature of a diffusion pattern may become low. In addition, when the voice section is divided between the voiced section and non-vocal register, in the voiced section, the noise nature of a diffusion pattern is low, and between non-vocal register, it is carried out so that the noise nature of a diffusion pattern may become high. The classification of the noise section and the voice section (between the voiced section and non-vocal register) is separately performed by the mode judging machine 1713 etc., and the mode information outputted from the mode judging machine 1713 performs selection of a diffusion pattern with the diffusion pattern generation vessel 3603.
[0271] That is, the mode judged with the mode judging vessel 1713 is outputted to the diffusion pattern generation machine 3603 as mode information, and switches the noise nature of a diffusion pattern with the diffusion pattern generation vessel 3603 based on mode information. In this case, as shown in drawing 38 , with the diffusion pattern generation vessel 3603, two or more noise nature is set up corresponding to the mode, and the strength of noise nature is chosen according to the mode. In the case of noise mode, the noise nature of a diffusion pattern chooses a strong thing, and, specifically, in the case of voice (voiced) mode, the noise nature of a diffusion pattern chooses a weak thing.

[0272] Moreover, with another diffusion pattern generation vessel 3603 of a configuration, a diffusion pattern performs actuation equivalent to the aforementioned change continuously by changing the amplitude value of 1 sample eye of a diffusion pattern corresponding to the height of the average power of the noise section. When the average power of the noise section is large, the multiplier which makes amplitude value of eye one sample small takes an advantage, and as shown in drawing 39 , when the average power of the noise section is small, specifically, the multiplier which enlarges amplitude value of eye one sample takes an advantage. About these multipliers, a transform function and the conversion Ruhr are beforehand appointed at the ability to determine using the value of the average power of the noise section. In addition, about the sample which changes amplitude value, it is not limited to one sample. Moreover, the diffusion pattern after multiplying by the multiplier is normalized so that it may become the same BEKUTORUPAWA as the pattern before multiplying by the multiplier.
[0273] Next, in voice decryption equipment, the average power between noise divisions is computed with the average power calculation vessel 2501 between noise divisions using PAWA of the frame (factice) judged in the past to be the noise section. Past noise section PAWA is serially updated using PAWA outputted from the present PAWA calculation machine 2502. The average power of the noise section computed here is outputted to the diffusion pattern generation machine 3703. With the diffusion pattern generation vessel 3703, the noise nature of a diffusion pattern is switched based on the average power of the noise section. That is, as shown in drawing 38 , with the diffusion pattern generation vessel 3703, two or more noise level is set up corresponding to the height of the average power of the noise section, and noise nature is chosen according to the height of average power. When the average power of the noise section is large, the noise nature of a diffusion pattern chooses a high (strong) thing, and when the average power of the noise section is small, specifically, the noise nature of a diffusion pattern chooses a low (weak) thing.
[0274] Moreover, you may make it switch the noise nature of a diffusion pattern also in this case in the noise section and the voice section. In addition, the voice section may be further divided between the voiced section and non-vocal register. In this case, in the noise section, the noise nature of a change of a diffusion pattern is high, and it is performed in the voice section so that the noise nature of a diffusion pattern may become low. In addition, when the voice section is divided between the voiced section and non-vocal register, in the voiced section, the noise nature of a diffusion pattern is low, and between non-vocal register, it is carried out so that the noise nature of a diffusion pattern may become high. The classification of the noise section and the voice section (between the voiced section and non-vocal register) is separately performed by the mode judging machine 1810 etc., and the mode information outputted from the mode judging machine 1810 performs selection of a diffusion pattern with the diffusion pattern generation vessel 3703.

[0275] That is, the mode judged with the mode judging vessel 1810 is outputted to the diffusion pattern generation machine 3703 as mode information, and switches the noise nature of a diffusion pattern with the diffusion pattern generation vessel 3703 based on mode information. In this case, as shown in drawing 38 , with the diffusion pattern generation vessel 3703, two or more noise nature is set up corresponding to the mode, and the strength of noise nature is chosen according to the mode. In the case of noise mode, the strong thing of the noise nature of a diffusion pattern is chosen, and, specifically, in the case of voice (voiced) mode, the weak thing of the noise nature of a diffusion pattern is chosen.

[0276] Moreover, with another diffusion pattern generation vessel 3703 of a configuration, a diffusion pattern changes the noise nature of a diffusion pattern continuously by changing the amplitude value of 1 sample eye of a diffusion pattern corresponding to the height of the average power of the noise section. When the average power of the noise section is large, it multiplies by the multiplier which the amplitude value of 1 sample eye makes small, and as shown in drawing 39 , when the average power of the noise section is small, specifically, it multiplies by the multiplier to which the amplitude value of eye one sample becomes large. Between this multiplier and the average power of the noise section, the transform function and the conversion Ruhr which were appointed beforehand can intervene, and it can ask now for an amplitude-conversion multiplier from the information on average power. In addition, about the sample which changes amplitude value, it is not limited to one sample. Moreover, the diffusion pattern into which amplitude value was changed is normalized so that it may become the same BEKUTORUPAWA as the diffusion pattern before changing amplitude value.

[0277] About the change of the noise nature of the diffusion pattern by the average power of the noise section If two or more kinds are prepared and a diffusion pattern is switched combining the both sides of mode information and average noy ZUPAWA information using mode information Even when noy ZUPAWA is large, in the voice section (voiced section), it becomes possible to make noise nature of a diffusion pattern into below whenever [middle] etc., and the voice quality in a noise can be improved.

[0278] You may make it switch the noise nature of a diffusion pattern regardless of the height of PAWA of the noise section in the gestalt of this operation in the noise section and the voice section. In this case, like the above, in the noise section, the noise nature of a change of a diffusion pattern is high, and it is performed in the voice section so that the noise nature of a diffusion pattern may become low. In addition, when the voice section is further divided between the voiced section and non-vocal register, a change is performed in the voiced section so that it may be low and the noise nature of a diffusion pattern may become high between non-vocal register about the noise nature of a diffusion pattern.

[0279] In the gestalt 6 of the above-mentioned implementation, although the case where a good transformation partial algebraic-sign book / random sign book is used is explained, also when a common algebraic-sign book is used, it can apply in this invention.

[0280]

[Effect of the Invention] As explained above, according to this invention, at least two in two or more sound-source pulses generated from an algebraic-sign book can reduce the size of a noise sign book by generating only combination which approaches. The voice to digital converter and voice decryption equipment which made it possible to improve the quality over the silent section or the stationary noise section can be offered by storing a sound-source vector effective in the silent section or the stationary noise section in the part of the size reduced especially.

[0281] Moreover, in the system which performs carving [mode / the mode corresponding to the silent section or the stationary noise section, and / corresponding to the other part (for example, voiced section)], the voice to digital converter and voice decryption equipment which made it possible to raise more the improvement factor of quality to the silent section or the stationary noise section can be offered by changing said size to reduce accommodative.

[Translation done.]